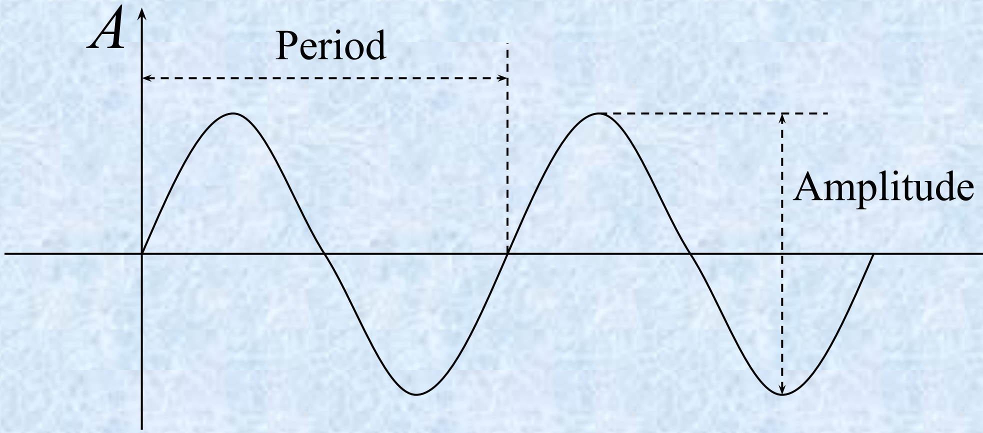
Chapter 2
Digital Audio Processing

NB. Please install Adobe Audition CC 2014 on your notebook

2.1 Basic of Digital Audio

Basic of Sound

Sound: a travelling wave that is an oscillation of pressure transmitted through a solid, liquid, or air, composed of frequencies within the range of hearing.

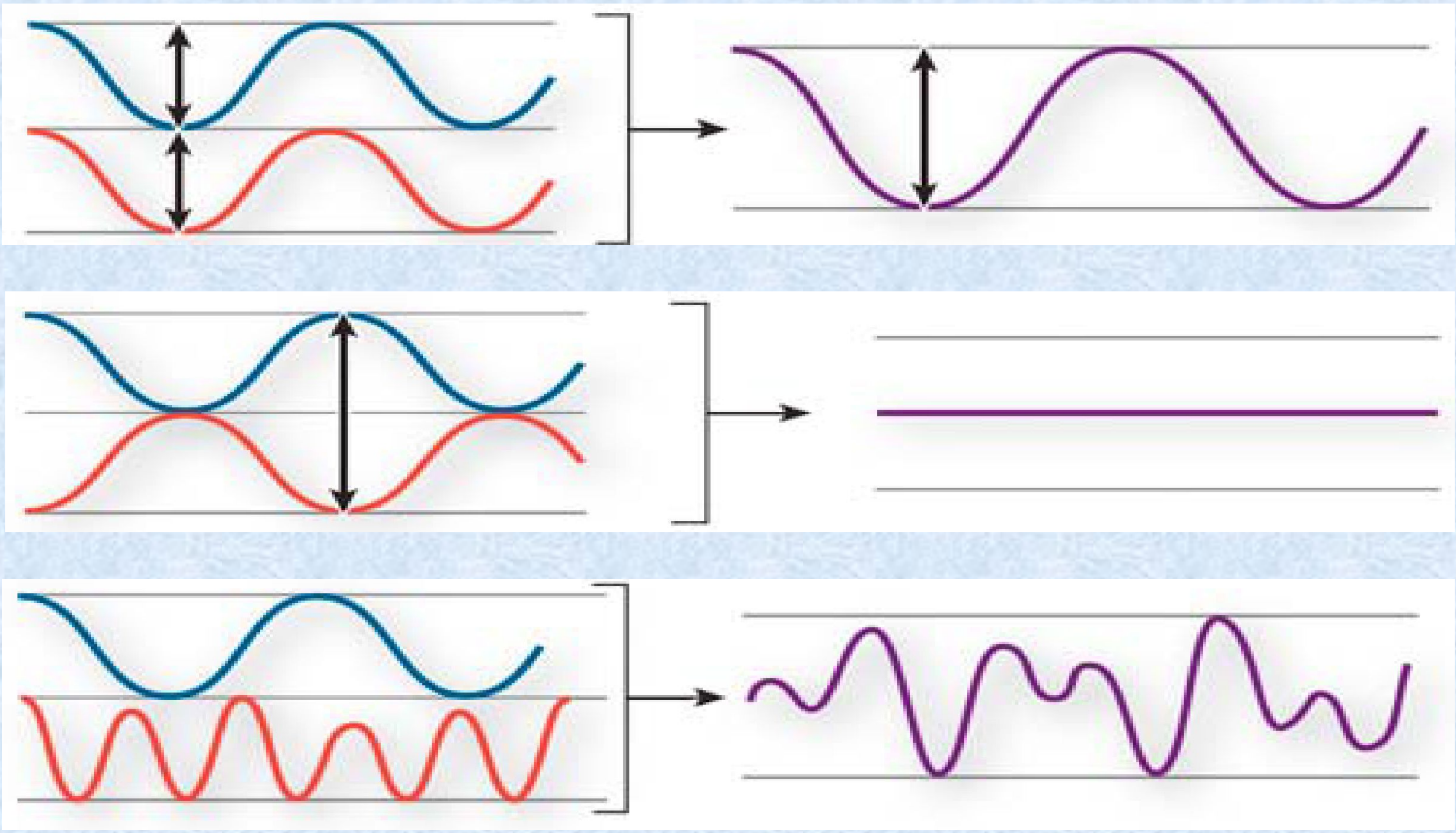


Essential Properties

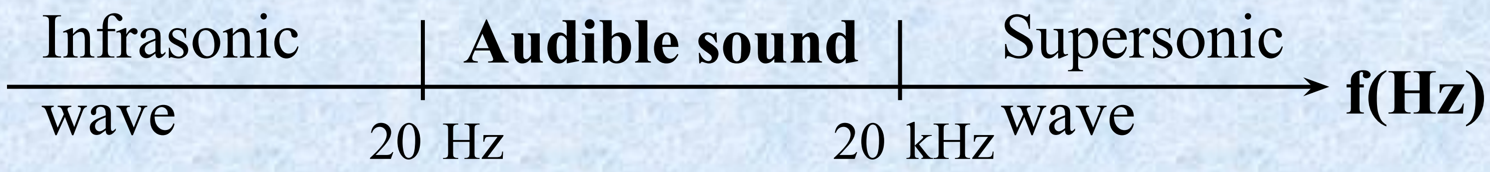
Amplitude: the magnitude of change during one oscillation.
Period: the time interval between two successive occurrences of a recurrent event.
Frequency: the number of occurrences of a repeating event per unit time.

How sound waves interact

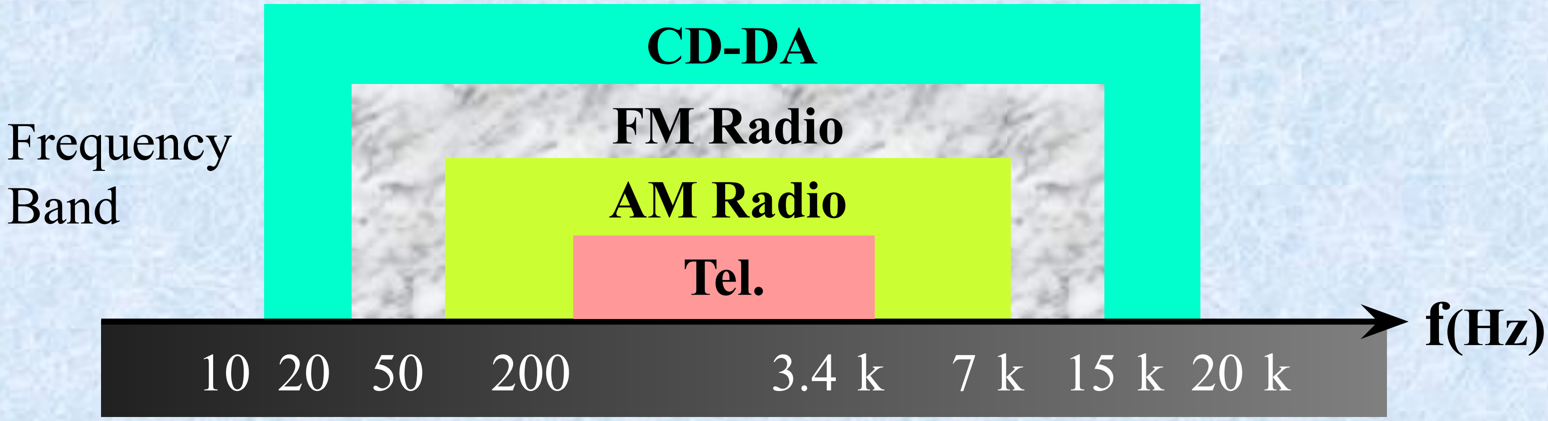
When two or more sound waves meet, they add to and subtract from each other.



Sound Frequency



Speech Signal Frequency：300Hz-3kHz

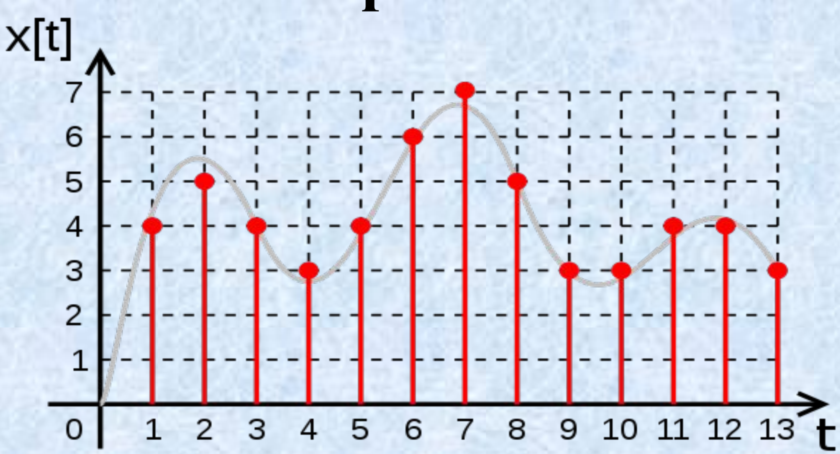


2.2 Audio Digitalization

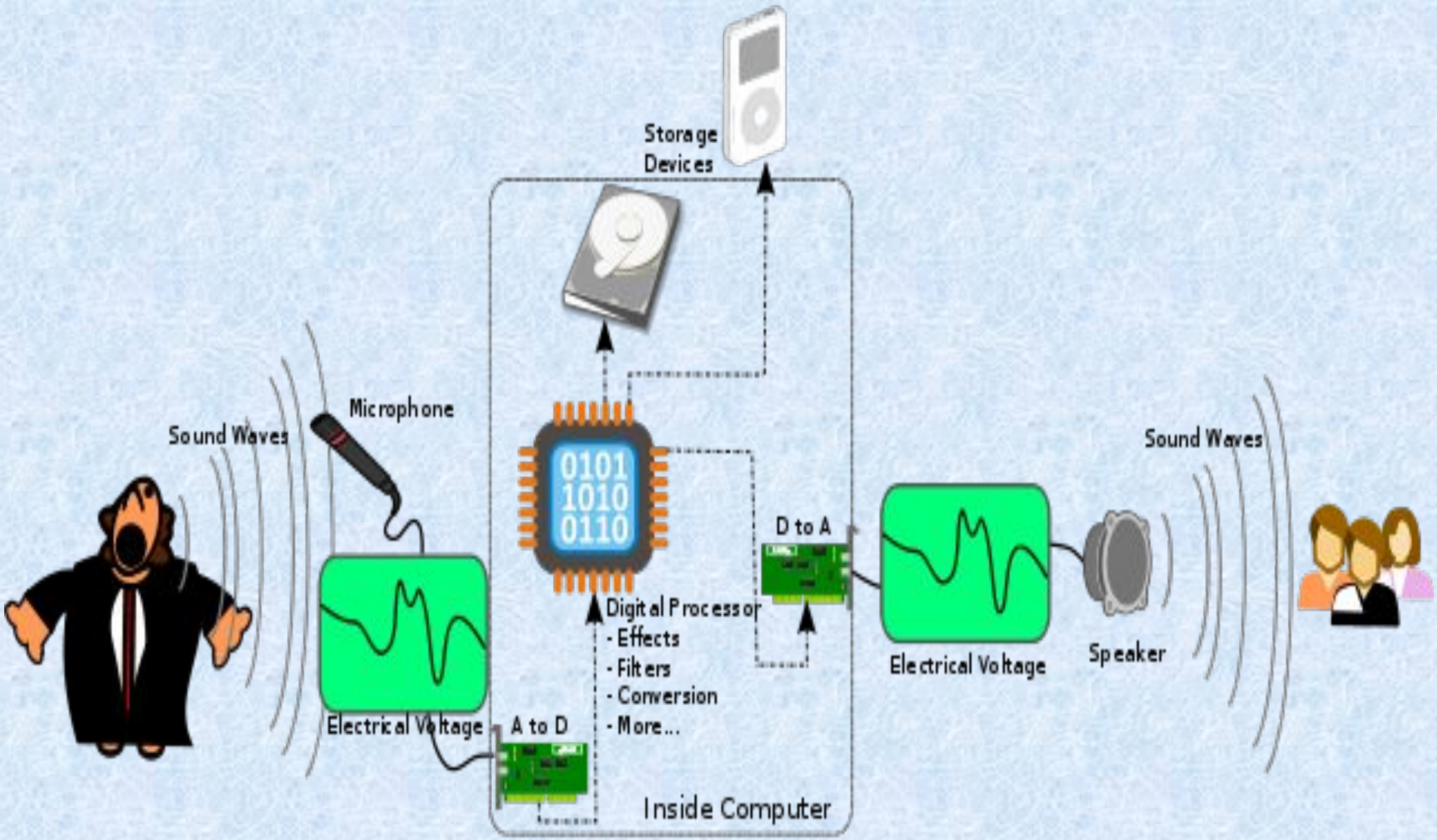
Audio signals

- **Analog Signal: an electrical representation of sound originated by microphone, tape head. Loudspeakers or headphones convert an electrical audio signal into sound.**

- **Digital Signal: a discrete-time signal for which not only the time but also the amplitude has discrete values**



Lifecycle of sound in Multimedia System

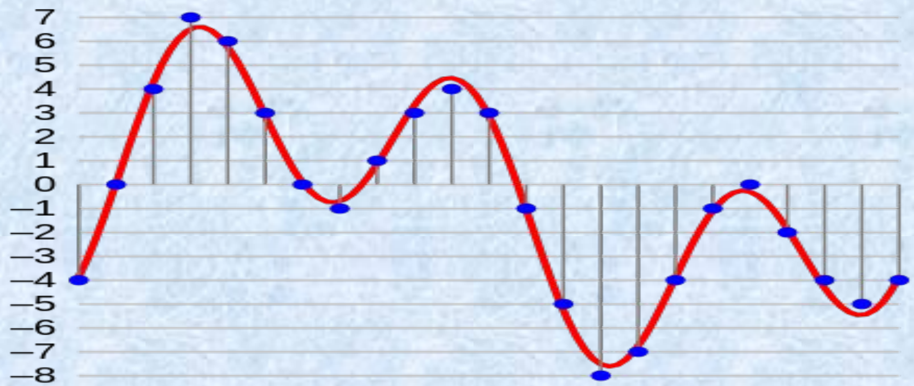


Procedure of audio digitalization

Analog Signal (Nature World) —A/D ADC→ **Digital Signal** (Computer)

Digital Signal —D/A DAC→ **Analog Signal**

Analog Signal → Sampling → Quantization → Coding → **Digital Signal**

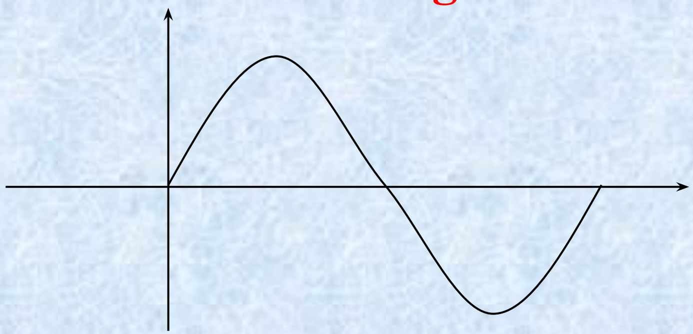


conversion to a stream of numbers, and preferably these numbers should be integers for efficiency.

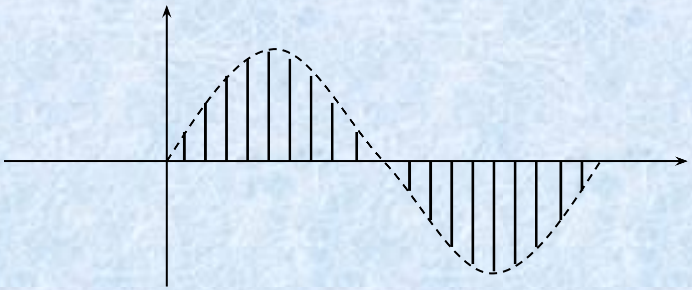
Procedure of audio digitalization

- **Sampling** means dividing continuous time into discrete time
- **Quantization** means measuring the amplitude at fixed interval
- **Coding** means forming a digital sequence in accordance with certain rules

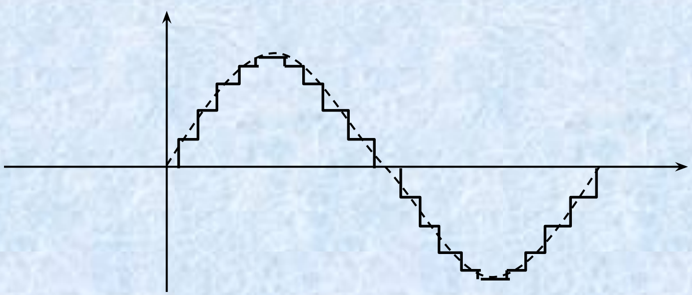
Continuous Analog Waveform



Sampling



Quantization



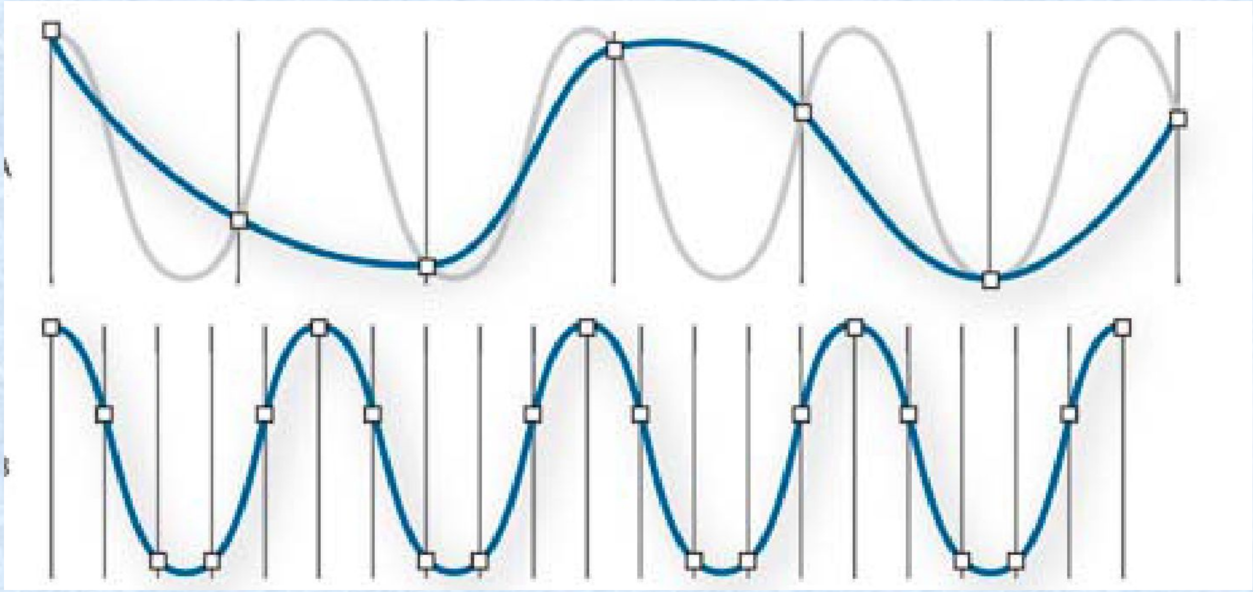
Discretized Audio

Illustration

Coding: 10100100…1110001000

Understanding sample rate

Sample rate indicates the number of digital snapshots taken of an audio signal each second. This rate determines the frequency range of an audio file. The higher the sample rate, the closer the shape of the digital waveform is to that of the original analog waveform.



Understanding sample rate

To reproduce a given frequency, the sample rate must be at least twice that frequency. For example, CDs have a sample rate of 44,100 samples per second, so they can reproduce frequencies up to 22,050 Hz, which is just beyond the limit of human hearing, 20,000 Hz.

Sample rate	Quality level	Frequency range
11,025 Hz	Poor AM radio (low-end multimedia)	0–5,512 Hz
22,050 Hz	Near FM radio (high-end multimedia)	0–11,025 Hz
32,000 Hz	Better than FM radio (standard broadcast rate)	0–16,000 Hz
44,100 Hz	CD	0–22,050 Hz
48,000 Hz	Standard DVD	0–24,000 Hz
96,000 Hz	Blu-ray DVD	0–48,000 Hz

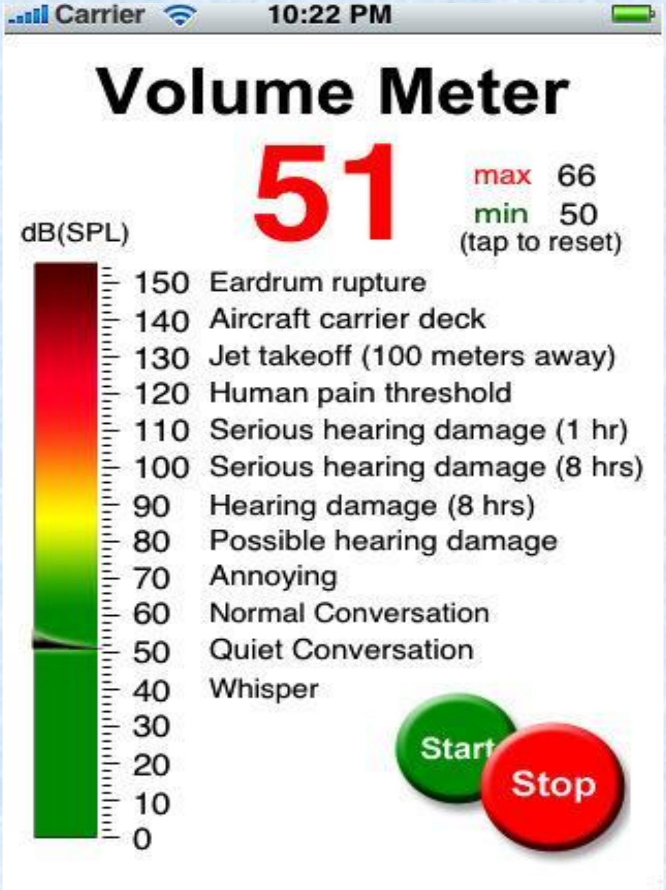
Understanding bit depth

When a sound wave is sampled, each sample is assigned the amplitude value closest to the original wave's amplitude. Higher bit depth provides more possible amplitude values, producing greater dynamic range, a lower noise floor, and higher fidelity.

Bit depth	Quality level	Amplitude values	Dynamic range
8-bit	Telephony	256	48 dB
16-bit	Audio CD	65,536	96 dB
24-bit	Audio DVD	16,777,216	144 dB
32-bit	Best	4,294,967,296	192 dB

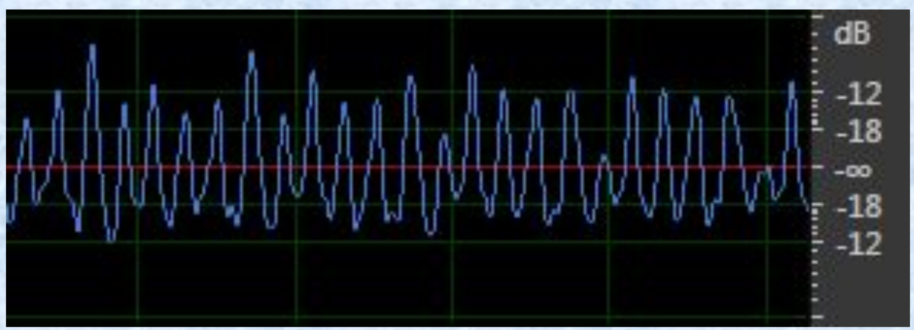
dB(Decibel)

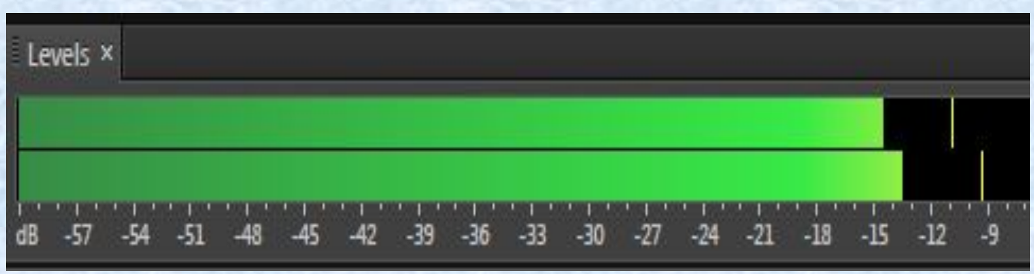
dB is commonly used in acoustics as a unit of sound volume. 0dB is the quietest sound our ear can hear, 120dB will cause permanent damage to our ear.



scale)

dBFS(dB relative to full scale) is a unit of measurement for amplitude levels in digital systems. The level of 0 dBFS is assigned to the maximum possible digital level.





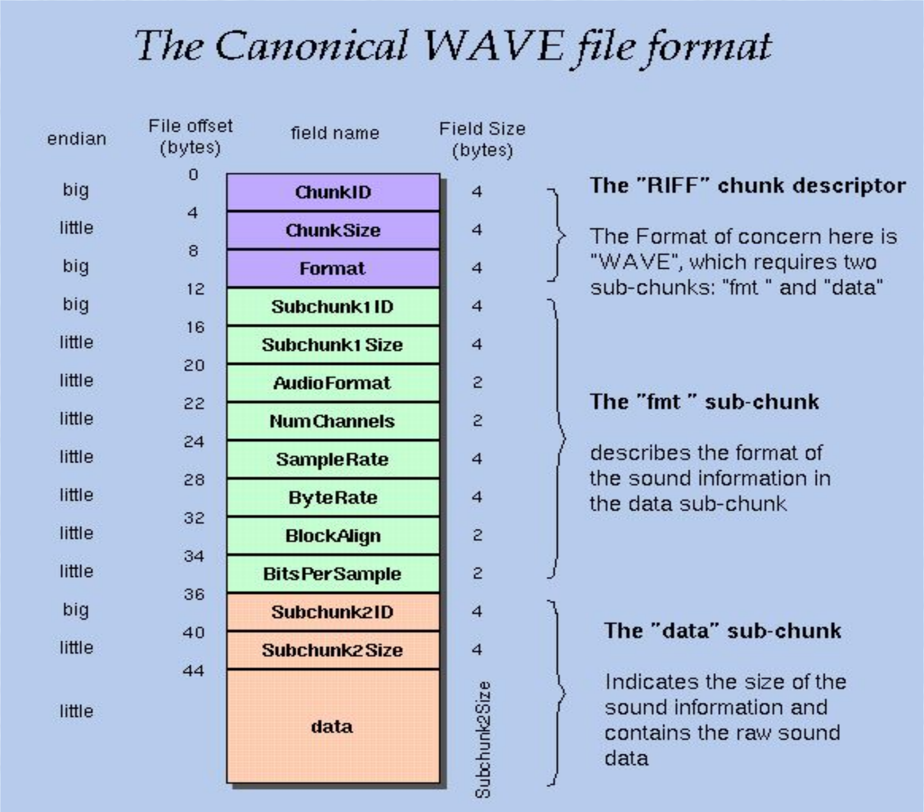
Affects of Data Size

Data Size =Sample Rate×Bit Size×#Channel/8(Byte/s)

Sample Rate (kHz)	Bit Size (bit)	Data Size(KB/s)	
		Mono	Stereo
11.025	8	10.77	21.53
	16	21.53	43.07
22.05	8	21.53	43.07
	16	43.07	86.13
44.1	8	43.07	86.13
	16	86.13	172.27

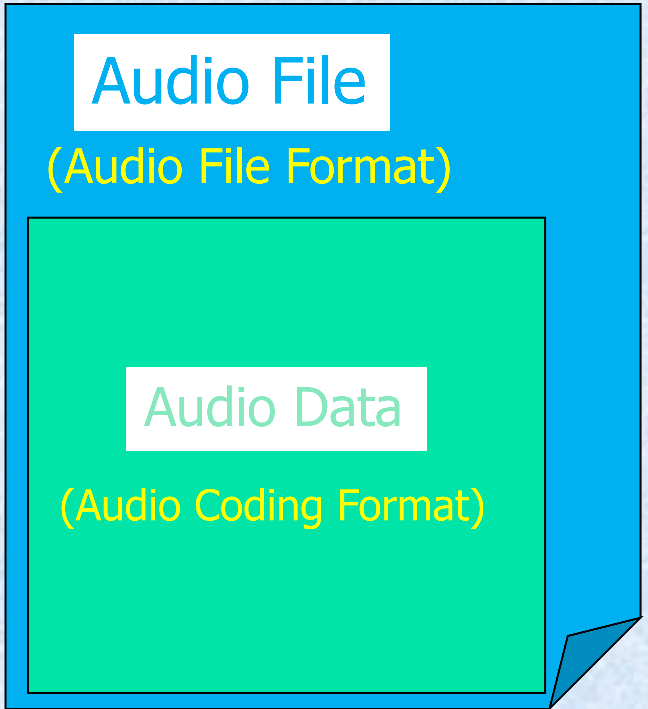
2.3 Audio File Format

- An audio file format is a file format for storing digital audio data on a computer system.
- An audio file usually contains a header indicating sample rate, bit depth etc(metadata), and then a large number of digital audio data.



Format types

- The bit layout of the audio data (excluding metadata) is called the audio coding format and can be uncompressed.
- It is important to distinguish between audio file(container) format and audio coding format.
- A coding format and a file format are usually defined in one compression standard, so most audio file formats support only one type of audio



Typical Audio File Formats

*.WAV Standard audio file container format used mainly in Windows PCs. Commonly used for storing uncompressed (PCM), CD-quality sound files (large in size). (see music.wav)

*.MP3 Defined in MPEG-1 Audio Layer-3. It is the most common sound file format used today.(Not to be confused with MPEG-3.)

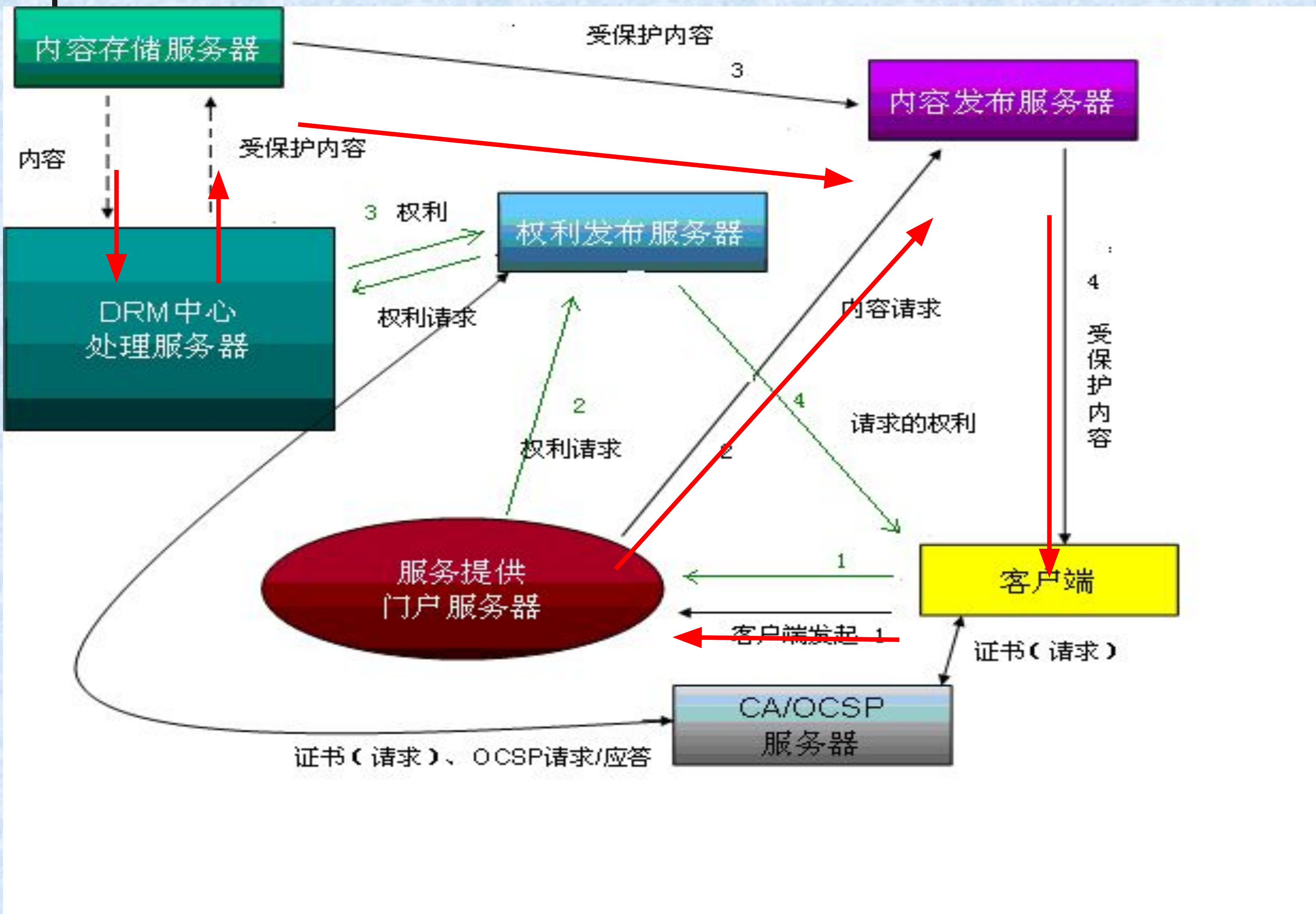
*.AAC Part of MPEG-2 and MPEG-4. Designed to be the successor of the MP3 format. Default file format for YouTube, iPhone, iPod, iPad, iTunes etc.

*.WMA(Windows Media Audio) owned by Microsoft. Designed with Digital Rights Management (DRM) abilities for copy protection.

*.OGG A patent-free, open source container format supporting a variety of coding format

*.APE For a lossless audio compression format. Take up several times as much space as lossy compression formats. (see CDImage.ape)

DRM Dataflow Diagram



2.4 Digital Audio Acquisition and Processing

Audio Acquisition

i) Sound Recorder Software(eg. Sound Recorder in Win 7)
ii) Recording Studio
iii) Audio CD, Audio Tape(eg. Window Media Player>Ripping)
iv) Digital Audio Library

Audio Processing

i) Audio Edition, Mixing
ii) Noise Reduction
iii) Modulation, Delay, Echo Effects

2.5 Digital Audio Compression Standards

Introduction to Digital Audio Compression

Compression Encoding: the process of encoding information using fewer bits than the original representation would use.

Audio compression relies on the facts:

- Information Redundancy in Sound.
- Human auditory system is not accurate within the width of a critical band (perceived loudness and audibility of a frequency).

Categories of Audio Compression Standards

Lossless Compression

- less space without losing any information.
- compression ratio of about 2:1

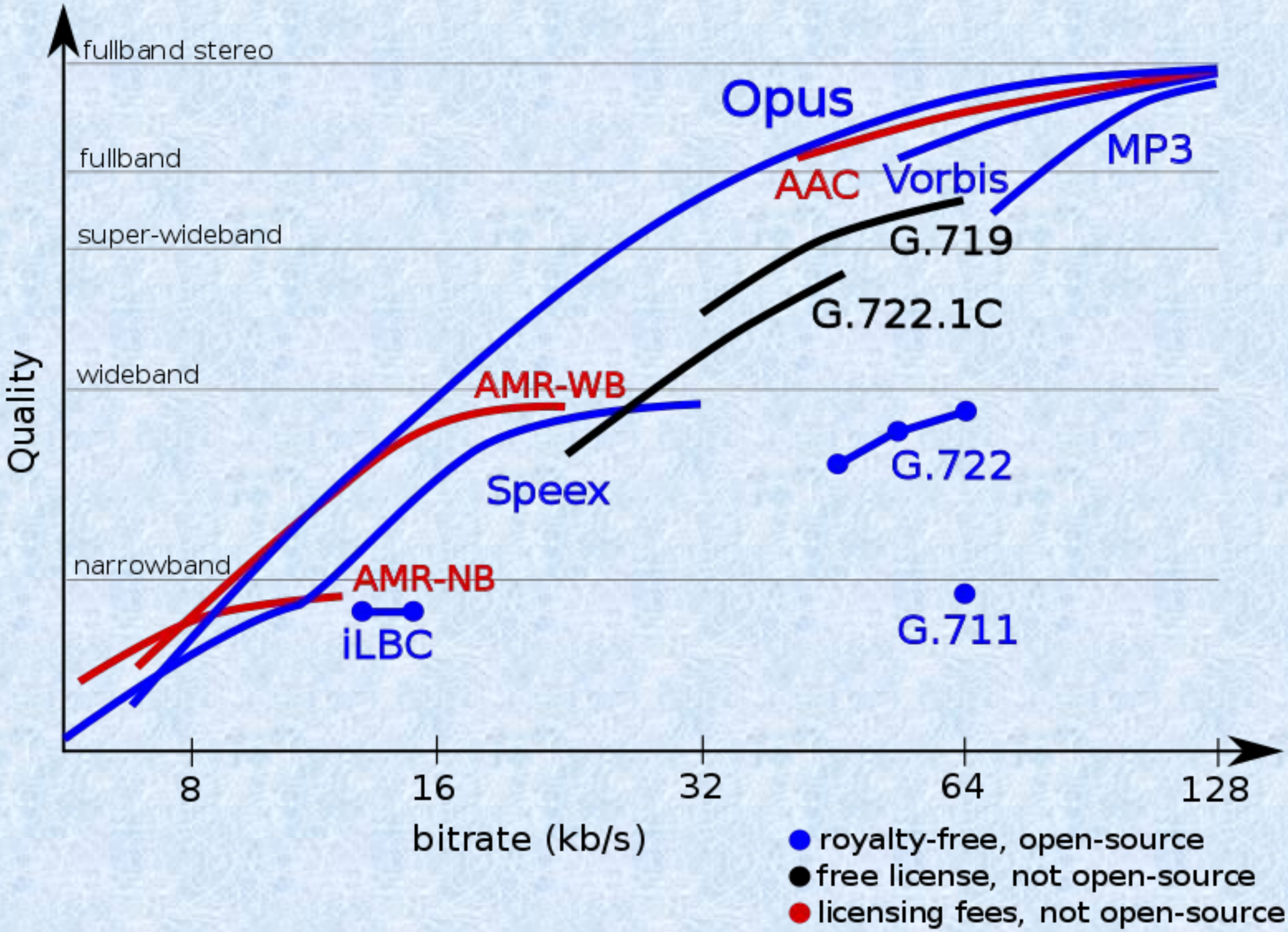
Lossy Compression

- greater reductions in file size
- reduction in audio quality
- most standards offer a range of degrees of compression, generally measured in bit rate

Audio Compression Standards

Categories	ITU Standard	Description
Telephony 200-3400Hz	G.711	Sample Rate: 8kHz, Bit Size: 8bit, Bit Rates: 64kb/s
	G.721	ADPCM , Bit Rates: 32 kb/s
	G.723	ADPCM Lossy, Bit Rates: 24 kb/s
	G.728	LD-CELP, Bit Rates:16 kb/s
AM Broadcasting 50-7000Hz	G.722	Sample Rate:16 kHz, Bit Size: 14bit, Bit Rates: 224(64) kb/s
Hi-Fi Stereo	MPEG Audio	Sample Rate: 44.1kHz, Bit Size: 16 bit, Bit Rates:705 kb/s (MPEG Layer-3, 384～64 kb/s)

Comparison between common audio formats



MP3

1993, MPEG-1 Audio Layer III

- a digital audio coding format which uses a form of lossy data compression
- more commonly referred to as MP3
- designed by the Moving Picture Experts Group (MPEG) as part of its MPEG-1 standard and later extended in the MPEG-2 standard.
- Compared to CD quality digital audio(44kKz, 16bit), MP3 compression commonly achieves 75 to 95% reduction in size.
  - CD:1.4Mb/s, MP3:128Kb/s

2.6 Sound Card and Electroacoustic Equipment

Sound Card

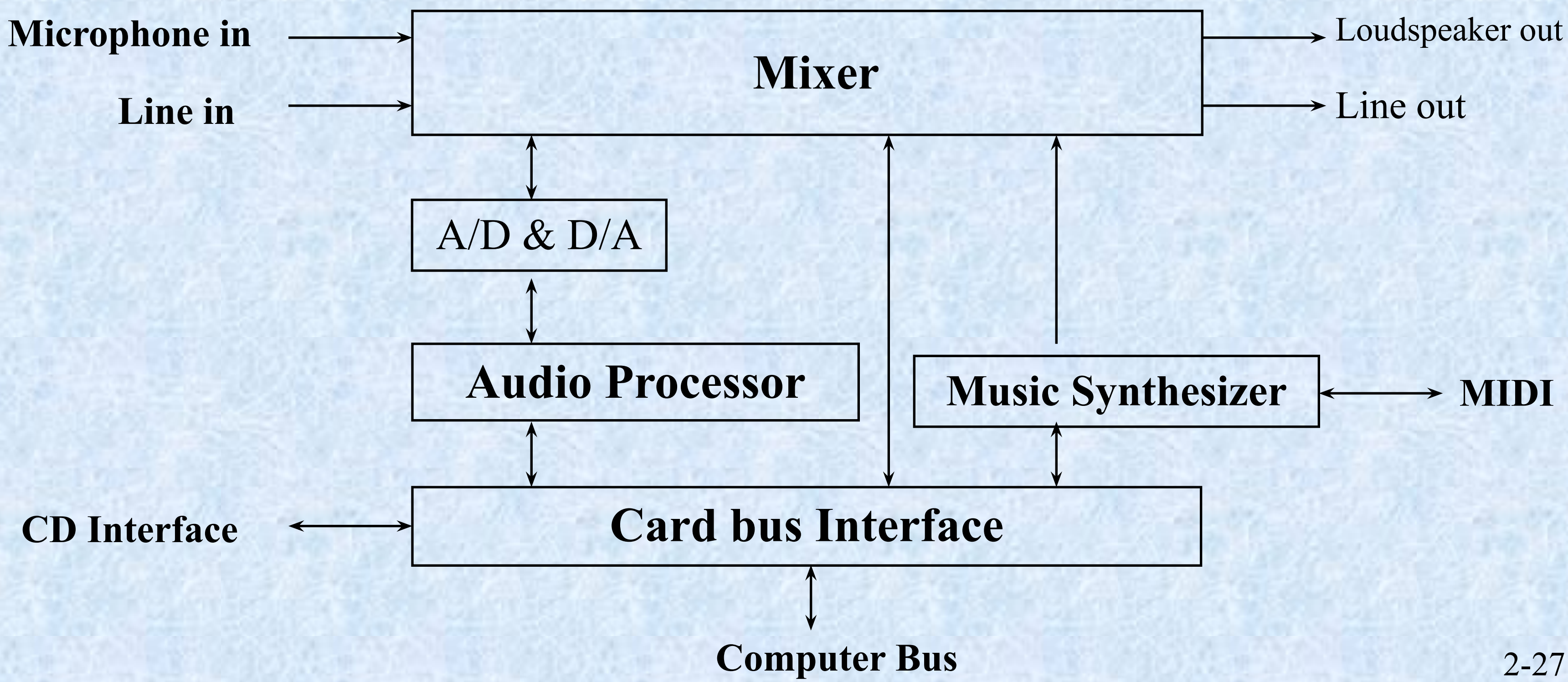
- An internal expansion card that facilitates input and output of audio signals to and from a computer under control of computer programs..

Integrated sound hardware on PC motherboards

- In the late 1990s many computer manufacturers began to replace plug-in soundcards with a "codec" chip integrated into the motherboard.
- The integrated sound system is often still referred to as a "sound card".
- The best plug-in cards, which use better and more expensive components, can achieve higher quality than integrated sound.

Architecture of Sound Card

Mixer receives inputs from both external connectors and D/A. It selects or mutes, amplifies these signals, adds them together, and finally routes the result to both external output connectors and A/D. (see Mixer in Win7)



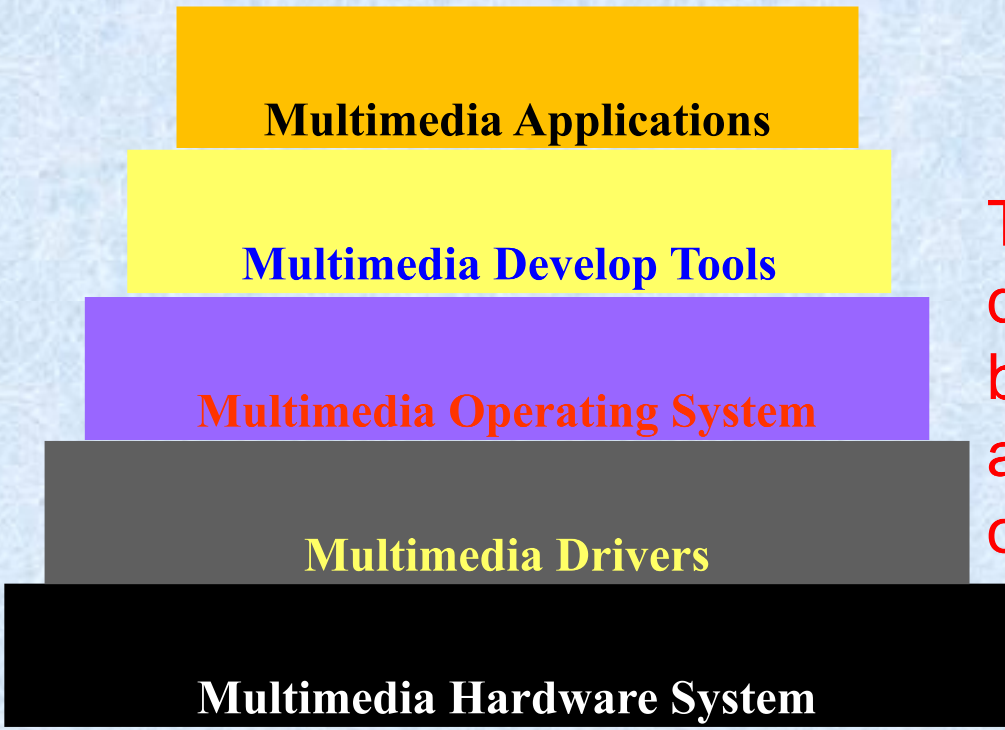
Audio Codec

A single chip in soundcard

- encodes analog audio as digital signals and decodes digital back into analog.
- compress and decompress digital audio data according to a given audio coding format. (may used by audio processing software and multimedia players)

Device Driver

- A low-level program that controls a device attached to a computer.
- Provides a software interface to hardware devices, enable operating systems to access hardware functions without needing to know details of the hardware being used.



The main purpose of Device drivers is to provide abstraction by acting as translator between a hardware device and the operating systems that use it

General Characteristics

i) Recording, editing, and playback of digital audio file.
ii) Controlling and mixing sounds from different sources.
iii) Compression and decompression in recording and playback.
iv) Music Synthesis.
v) Support MIDI interface.

It can be used in multimedia applications such as music composition, audio editing, presentation, education, entertainment (games) , etc

Example

- Enjoy multi-channel cinematic sound

Many sound cards come with 5.1 channel outputs so you can connect to your existing multi-channel speakers with ease.

Example

- ## Independent input sources

Many sound cards come with independent line-in and microphone connectors, which allow you to plug in two different audio sources to your PC.

You can plug in your MP3 player and sing along, while recording your singing session for your friends to enjoy!

Example

- Control and customize your audio

Many sound cards come with audio processing software, which is designed to bring cinematic difference to home theatre PCs! You can also adjust the level of immersion you like, simply by adjusting the sliders on Control Panel.

Quality Parameters

Audio Performance

Sample Rate:

11.025 kHz (Speech)

22.05 kHz (Music)

44.1 kHz (Hi-Fi)

Bit Size:

8 bits/256 (Speech)

16 bits/65 536 (Hi-Fi)

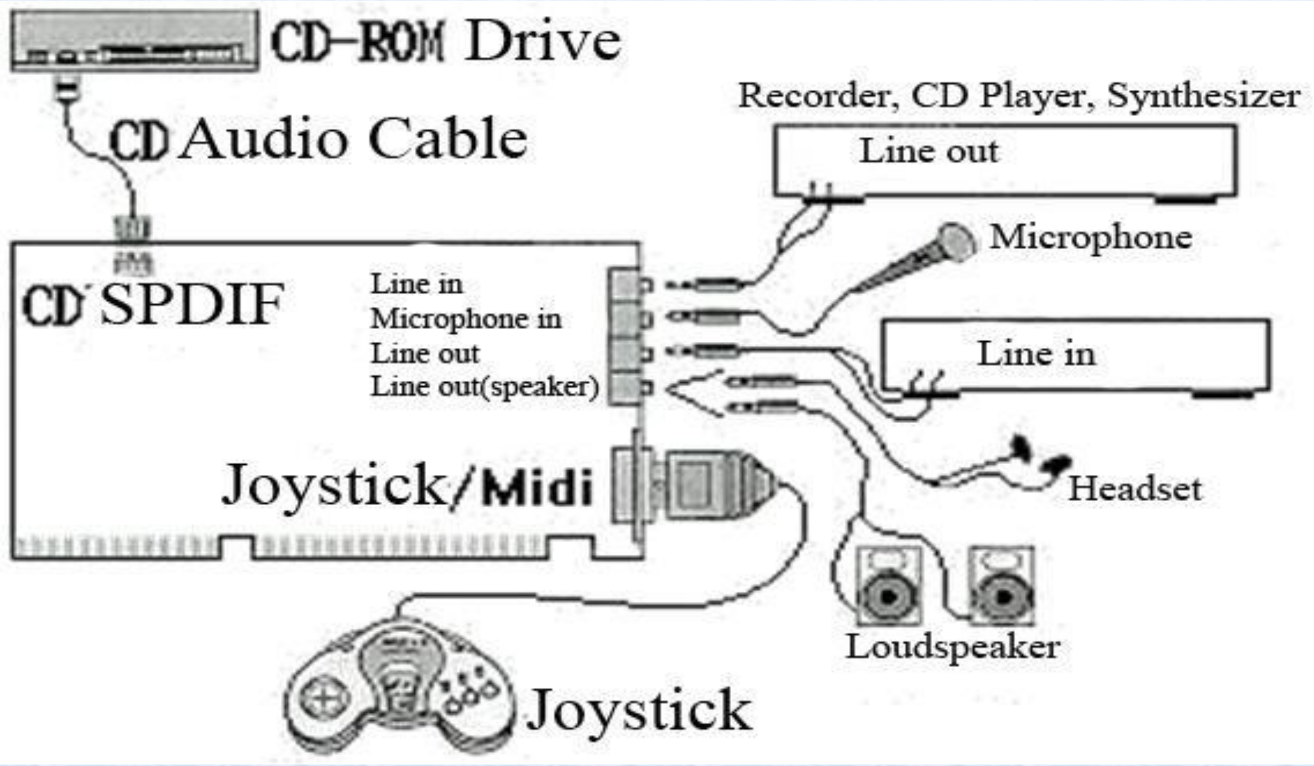
SNR(Signal-to-noise ratio):>80db

Quality Parameters

- **Connectivity**

  Input: **line in jack, microphone in jack**

  Output: **headphone, mono, 2 channels(Stereo)、2.1/4.1/5.1 Channels (Surround)**



Quality Parameters

Processor

CODEC (Dependent of CPU, Cheap)

DSP(Independent of CPU)

2.7 Electric Music and MIDI

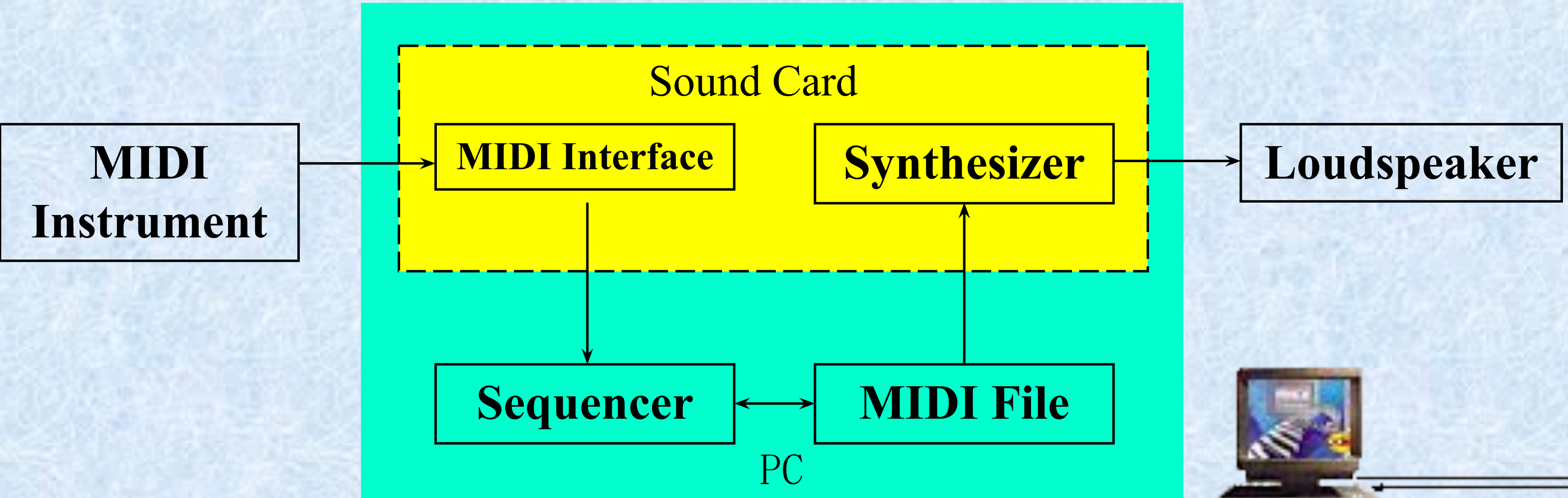
- **What is MIDI?**

MIDI (Musical Instrument Digital Interface): a technology to synthesize music using electronic equipment.

- **MIDI Standards**

An industry-standard that enables electronic musical instruments, computers and other electronic equipment to communicate and synchronize with each other.

MIDI Working Progress



There are actually three components in MIDI standard, which are the communications Protocol (language), the Connector (hardware interface) and a distribution format called Standard MIDI Files.

MIDI Protocol (language)

- **The MIDI protocol is an entire music description language in binary form like CPU machine language instructions for musical instruments. .**
- **MIDI language carries event messages that specify notation, pitch and velocity, control signals for parameters such as volume, vibrato, and etc.**

MIDI Connector

- **MIDI connector is a 5-pin DIN connector used to send MIDI messages .**
- **A single MIDI link can carry up to sixteen channels of information, each of which can be routed to a separate device. So MIDI allows multiple instruments to be played from a single controller which makes stage setups much more portable.**

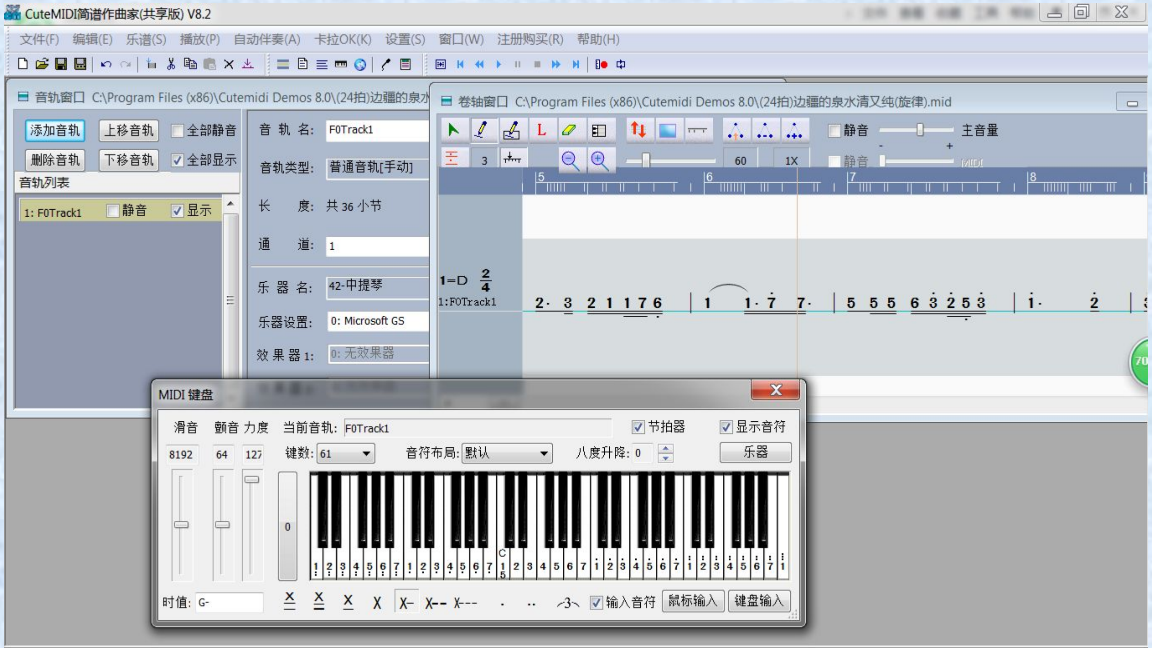
MIDI File

- **When MIDI messages are stored on disks, they are commonly saved in the Standard MIDI file format.**
- **MIDI file does not contain the actual sound, but only commands(like musical notation) to make the sounds, so it use a thousand times less disk space than the equivalent recorded audio.**

Example: midi_sample.mid

MIDI Sequencer

- Sequencer is the key component for MIDI music creation.
- A MIDI Sequencer (or simply sequencer) is a device or application software that can record, edit, or play back music, by handling note and performance information in MIDI



Example

MIDI is easy for modification and manipulation

Drum sample 1

Drum sample 2

Bass sample 1

Bass sample 2

A combination of the previous four files, with piano, jazz guitar, a hi-hat and four extra measures added to complete the short song, in A minor

2.8 Adobe Audition

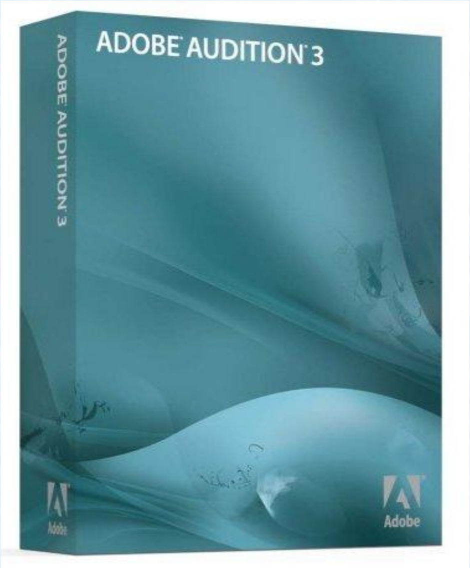
Audio Processing Software

(formerly Cool Edit Pro)

- Audio Editing
- Audio Effects
- Multitrack Processing
- Burning Audio CD

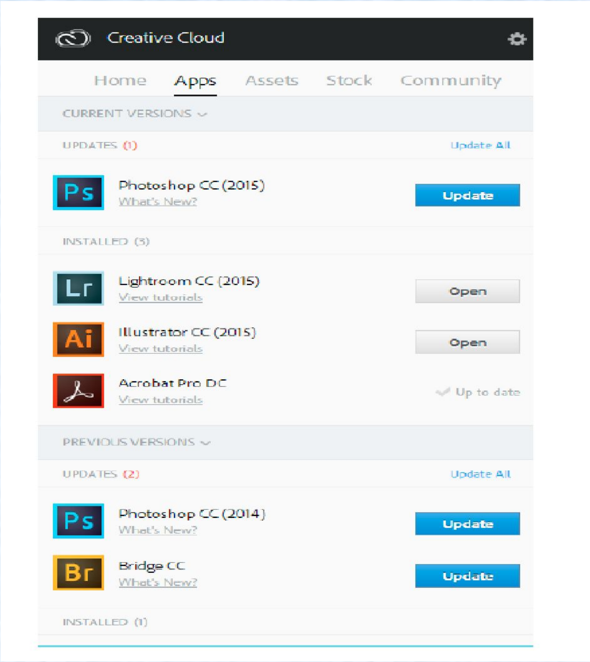
History

- 1990s, Syntrillium Software, Cool Edit,
- 2003, Adobe purchased Cool Edit Pro, Adobe Audition 1
- 2006, 2007 Adobe Audition 2, 3
- 2011, 2012 Adobe Audition 4, 5(CS5.5, CS6 as part of Adobe Creative Suite)
- 2013, 2014, 2015 Adobe Audition 6, 7, 8(CC, CC2014, CC2015)



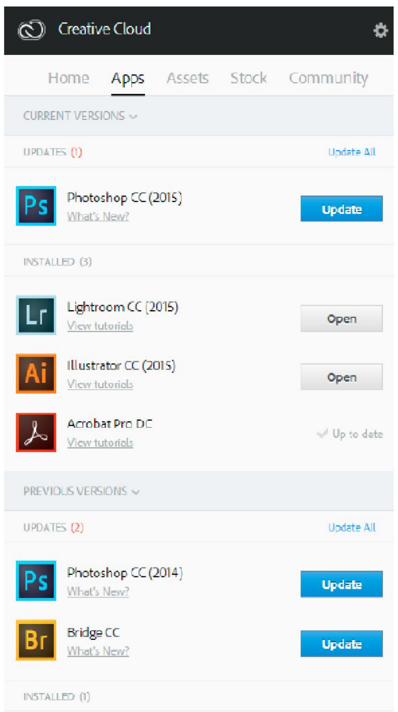
Creative Cloud

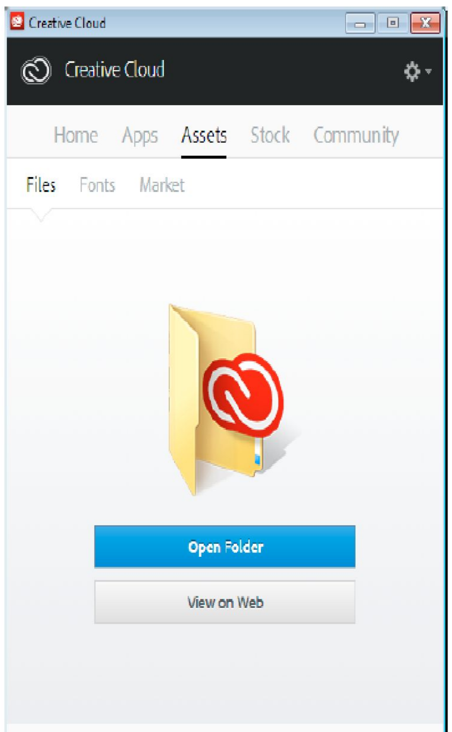
- **With the introduction of Creative Cloud branding, Adobe's licensing scheme was changed to that of software as a service and the "CS" suffixes were replaced with "CC".**
- **Adobe Creative Cloud allows licensed users to download, install, and update apps. You can also sync files and fonts, and showcase and discover creative work in community.**

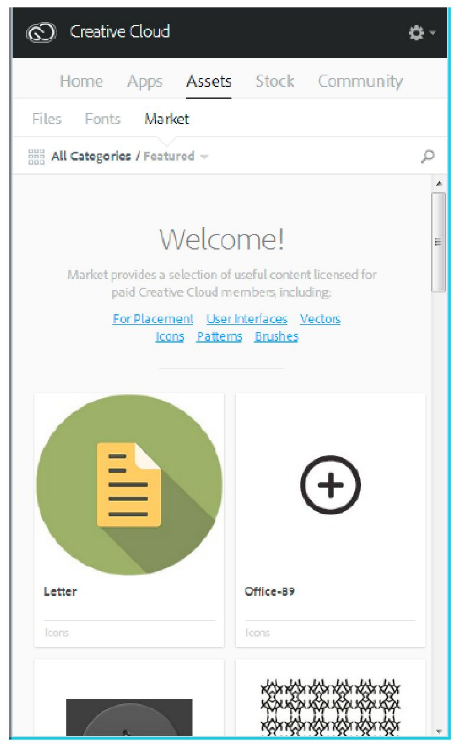


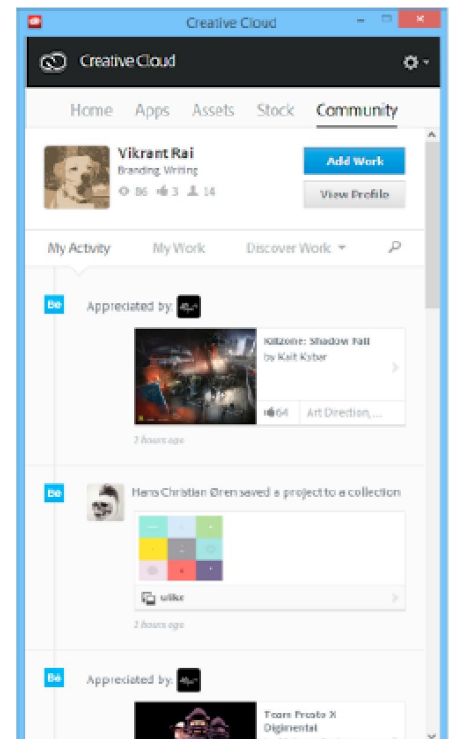
Creative Cloud for desktop

- Download and install apps
- Sync files and folders
- Add fonts
- Search for assets on Creative Cloud Market
- Share and discover work in community

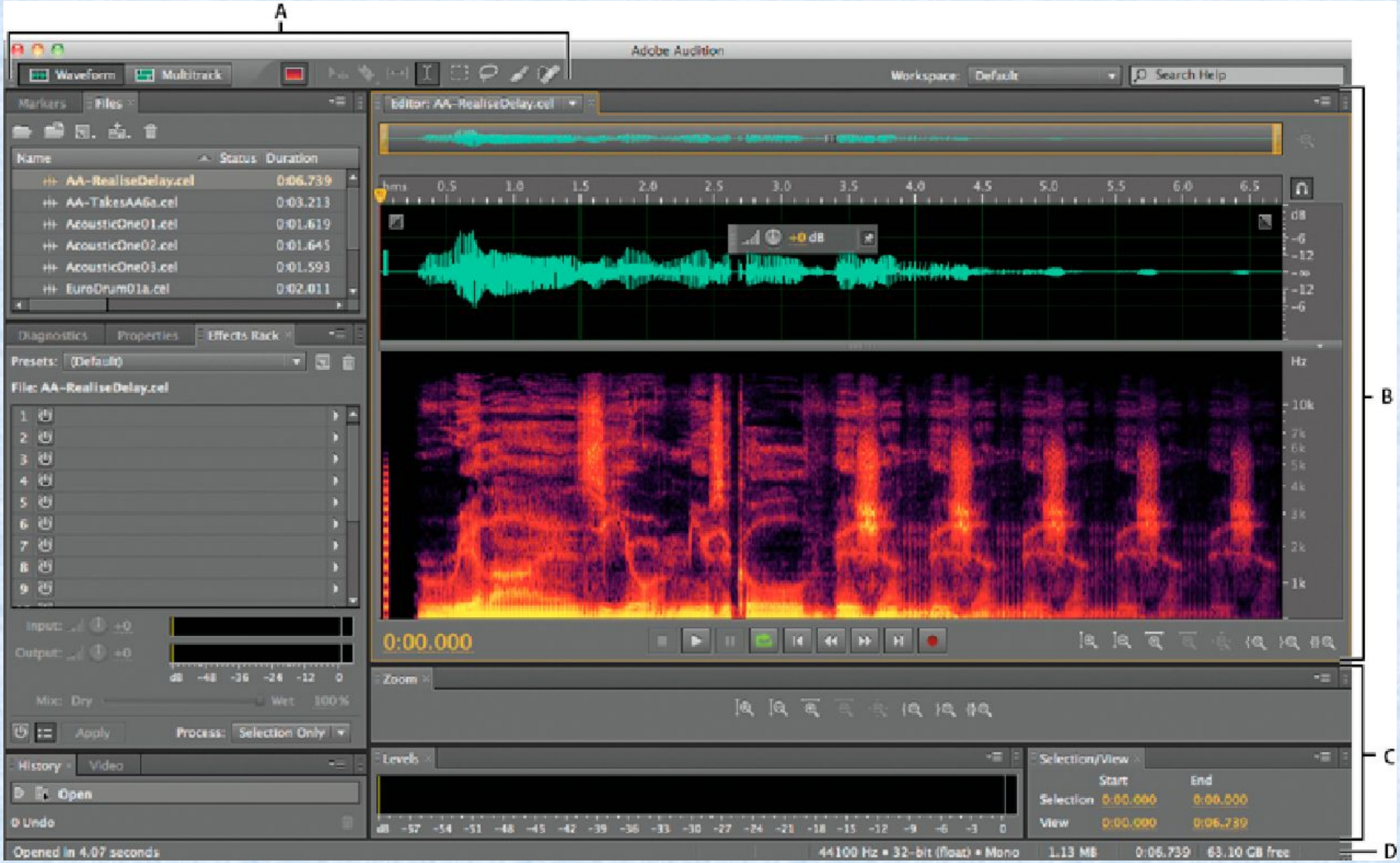








Workspace

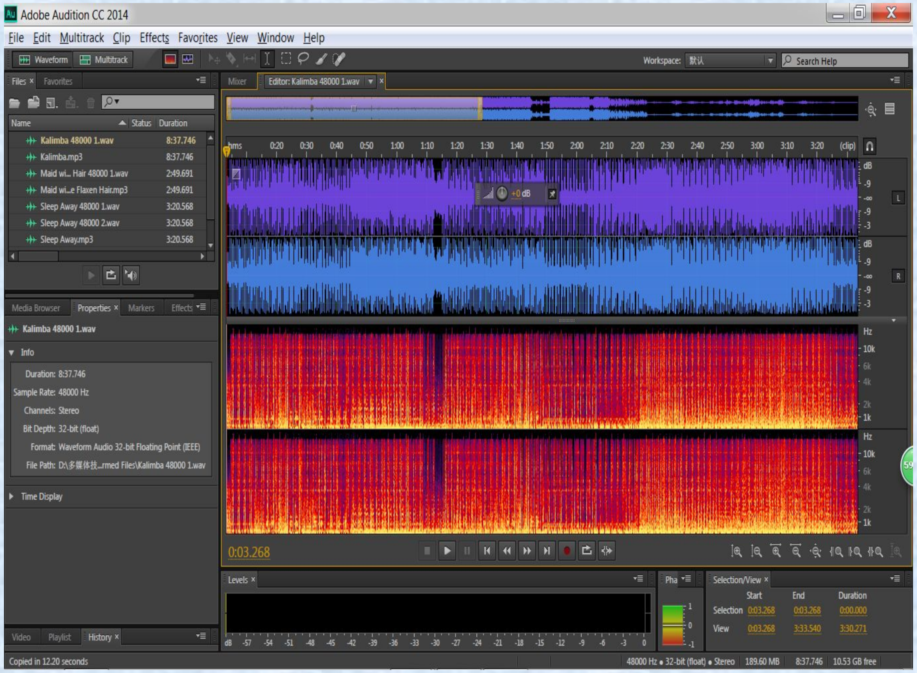


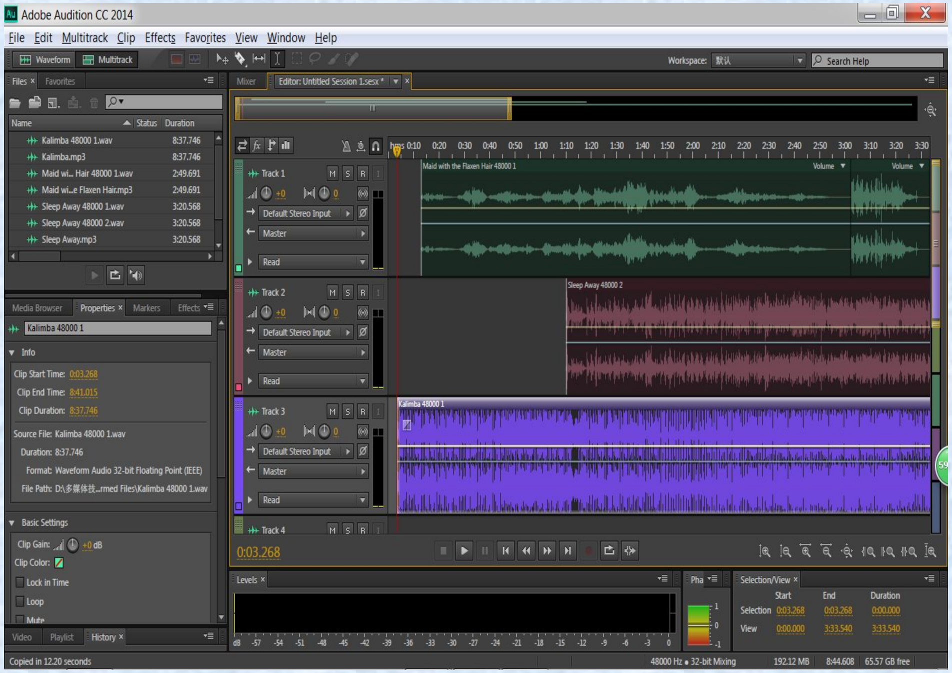
A *View buttons and toolbar* ***B*** *Editor panel with zoom navigator at top* ***C*** *Various other panels* ***D*** *Status bar*

Workspace

Waveform and Multitrack Editors

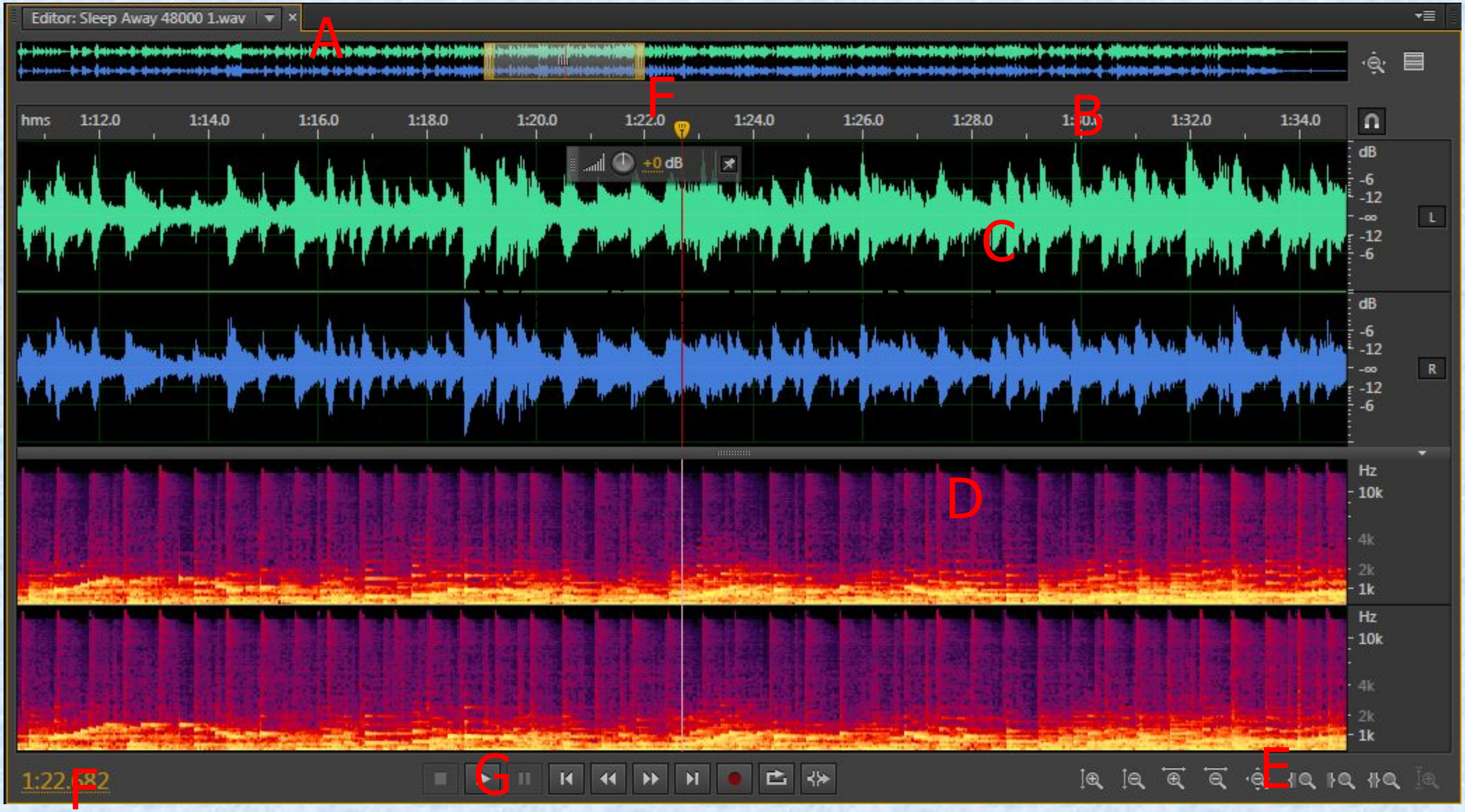
- To edit individual files, use the **Waveform Editor**. To mix multiple files and integrate them with video, use the **Multitrack Editor**.
- The **Waveform** and **Multitrack** editors use different editing methods, and each has unique advantages.





Workspace

Waveform Editor Panel



A *Navigator* ***B*** *Timeline Ruler* ***C*** *Waveform Display* ***D*** *Spectral Display*
E *Zoom Button* ***F*** *Current-Time Indicator/Display* ***G*** *Play Buttons*

Workspace

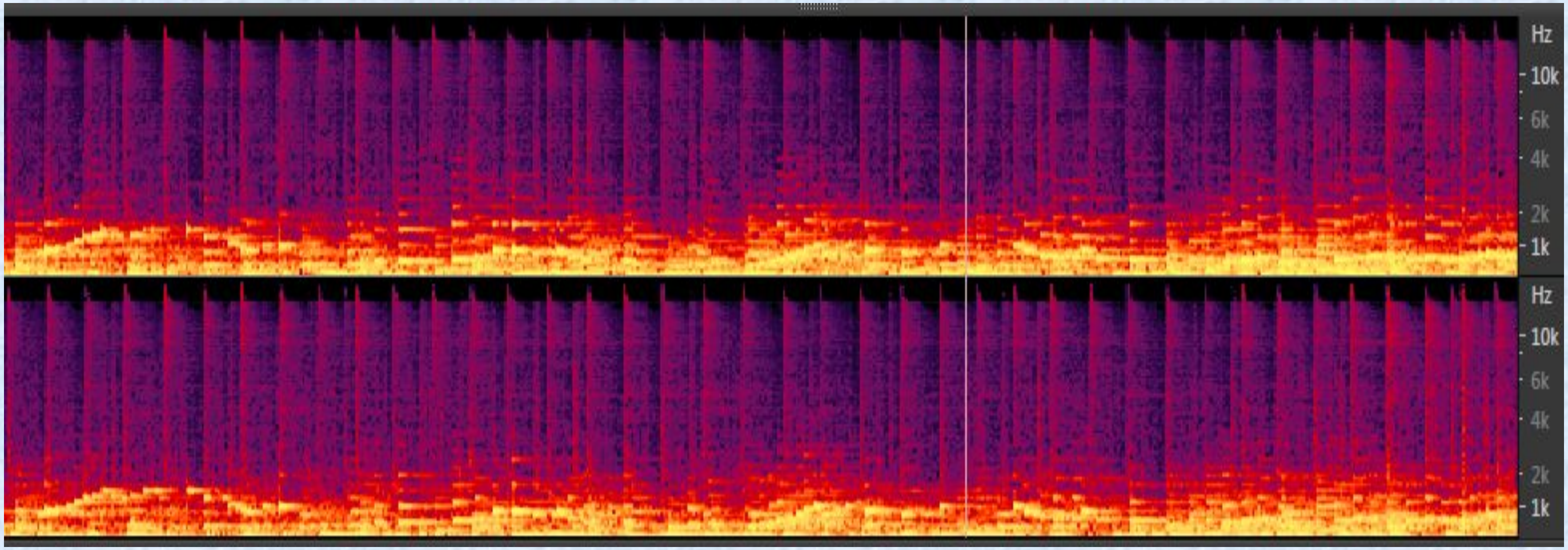
Waveform Display

- **Quiet audio has both lower peaks and lower valleys than loud audio.**
- **Channels can be viewed as layered or uniquely colored.**
    - View > Waveform Channels

Workspace

Spectral Display

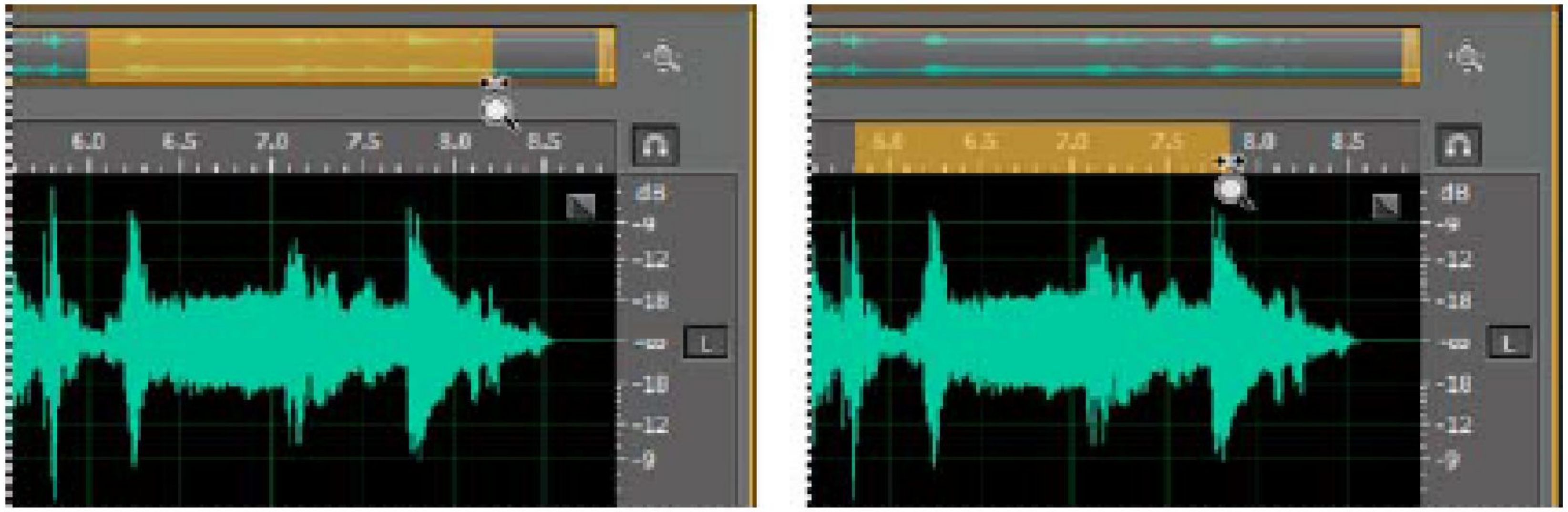
- This view lets you analyze audio data to see which frequencies are most prevalent. Brighter colors represent greater amplitude components. Colors range from dark blue (low-amplitude frequencies) to bright yellow (high-amplitude frequencies).
- The spectral display is perfect for removing unwanted sounds, such as coughs and other artifacts.



Workspace

Zoom audio

- **Zoom into a specific time range**
  - In either the zoom navigator or the timeline ruler, right-click and drag
- **Extend or shorten the displayed range**
  - Drag the left or right edge of the highlighted area in the zoom navigator
- **Using zoom button in the Editor panel**

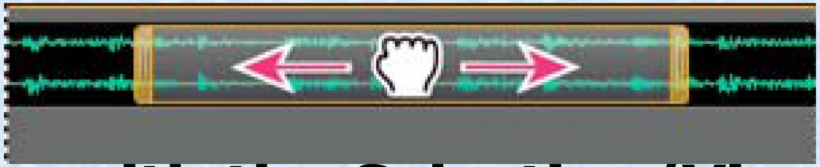


A Zoom navigator B Timeline ruler

Workspace

Navigating time and playing audio

- **Navigate by scrolling**



- **Navigate with the Selection/View panel**


Selection/View

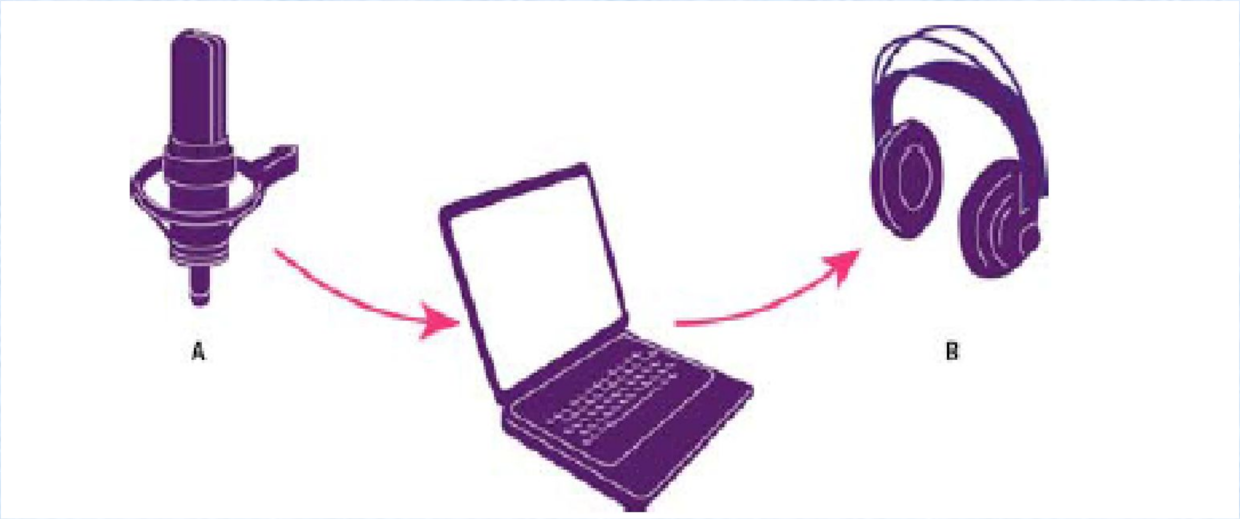
	Start	End	Duration
Selection	1:16.000	1:16.000	0:00.000
View	0:00.000	3:20.568	3:20.568

- **Current-time indicator(CTI) lets you start playback or recording at a specific point**

Importing, recording, and playing

Connecting to audio hardware

- **Configure audio inputs and outputs**
  - Choose Edit > Preferences > Audio Hardware
  - From the Device Class menu, choose the driver(eg. for the sound card)
  - Choose a Default Input and Output



Importing, recording, and playing

Creating and opening files

- **Create a new, blank audio file**
- **Open existing audio files**
- **Append audio files to another**
- **Extracting audio from CDs**

Importing with the Files panel

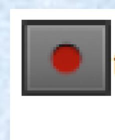
The Files panel displays a list of audio files for easy access

- **Import files into the Files panel**
- **Change displayed metadata in the Files panel**

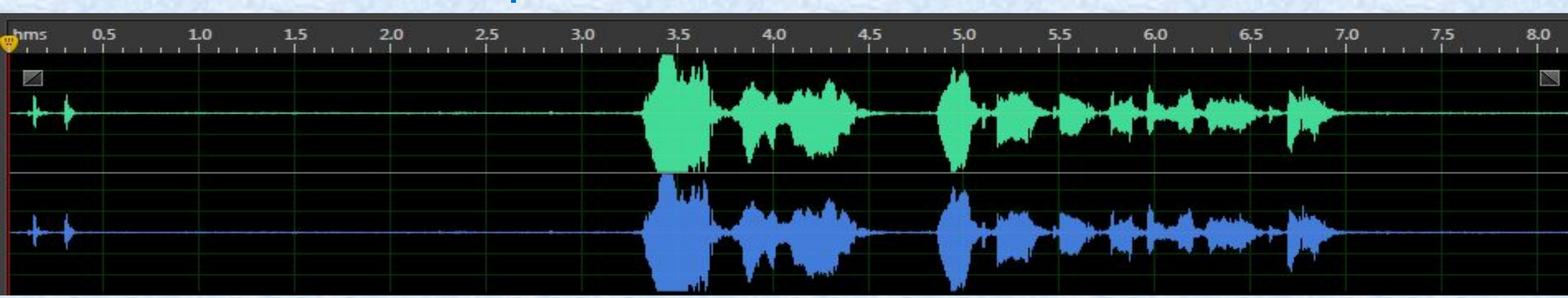
Importing, recording, and playing

Recording audio

You can record audio from a microphone or any device you can plug into the Line In port of a sound card.

- **Set audio inputs**
- **Create or open a file**
- **Click the Record button**  **to start and stop recording**

NB. When recording in noisy environments, record a few seconds of representative background noise that can be used as a noise print later on.



Importing, recording, and playing

Monitoring recording and playback levels

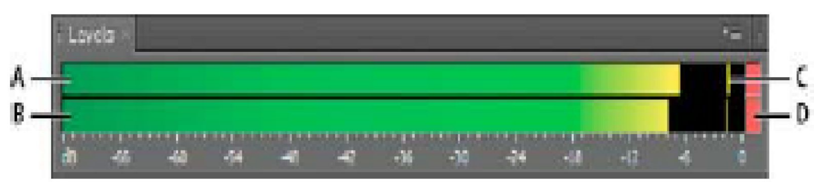
- To monitor the amplitude of incoming and outgoing signals during recording and playback, we use level meters.
- If amplitude is too low, sound quality is reduced; if amplitude is too high, clipping occurs and produces distortion.

A Left channel B Right channel C Peak indicators D Clip indicators

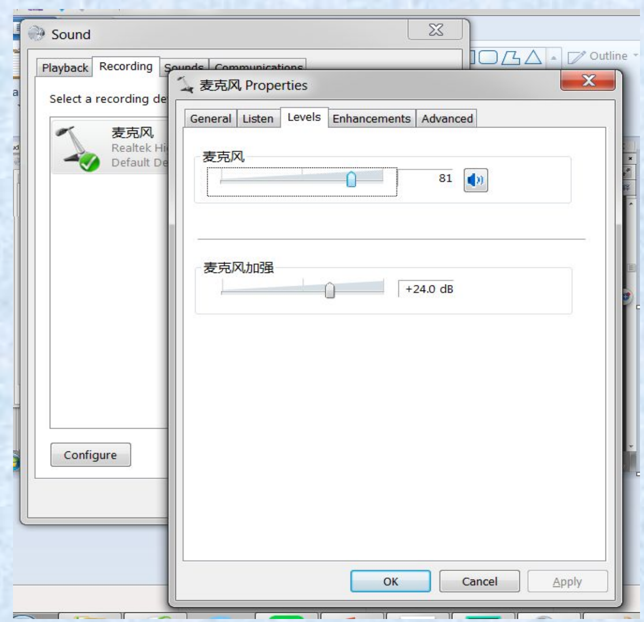
Importing, recording, and playing

Adjust recording levels for recording device

- Adjust levels if recordings are too quiet (causing unwanted noise) or too loud (causing distortion).
- To get the best sounding results, record audio as loud as possible without clipping.
- When setting recording levels, watch the meters, and try to keep the loudest peaks in the yellow range below -3 dB



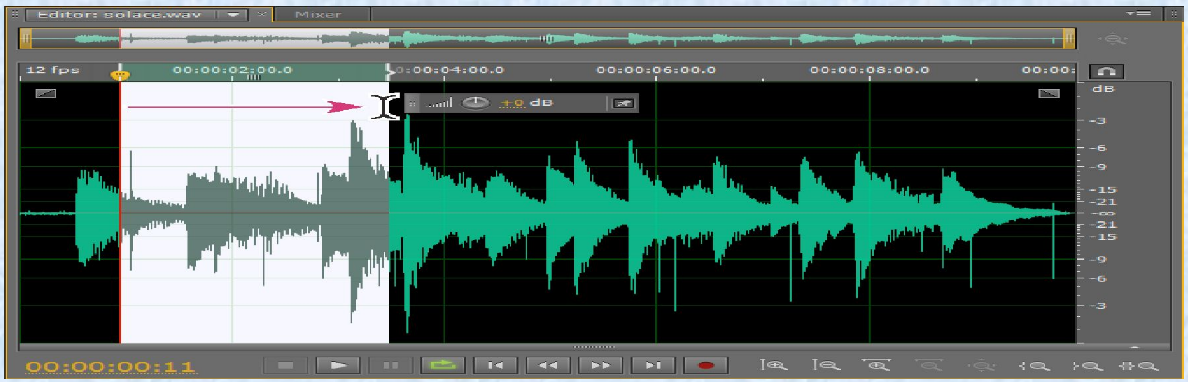
A Left channel B Right channel C Peak indicators D Clip indicators



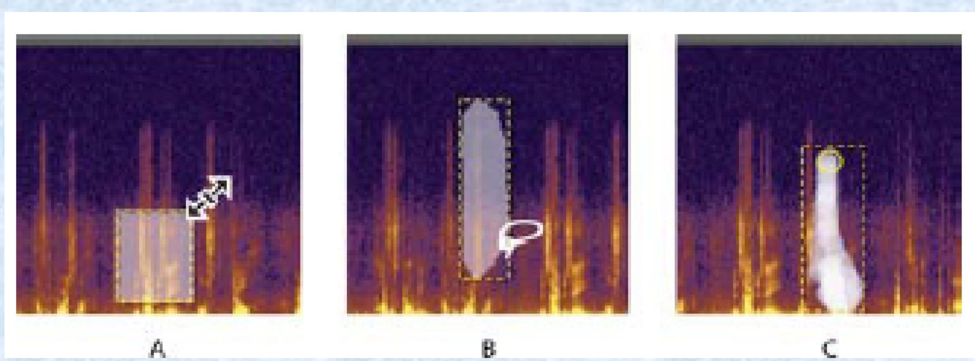
Editing Audio

Selecting audio for playing, copying, cutting, (mix)pasting, and deleting

- **Select time ranges using Time Selection tool** 



- **Select spectral ranges using free-form selection tools**



***A** Marquee*

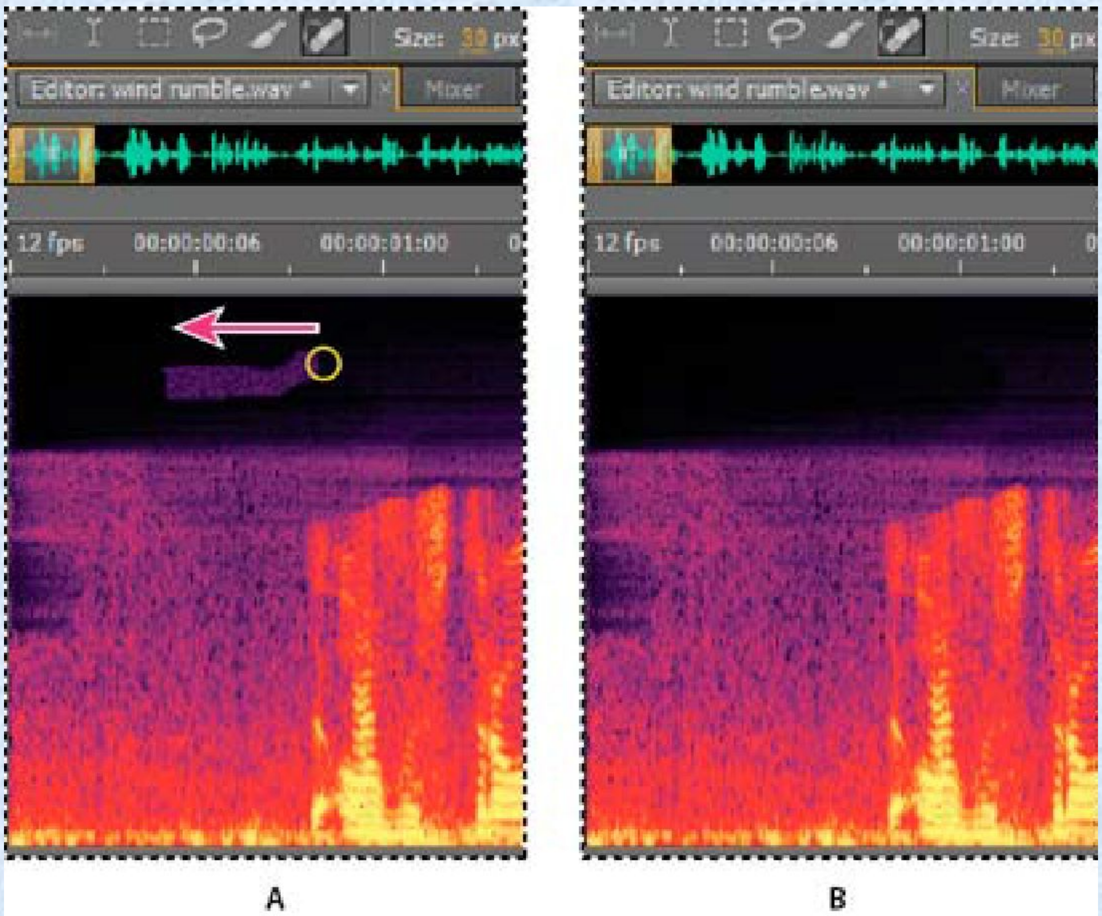
***B** Lasso*

***C** Paintbrush*

Editing Audio

Remove artifacts automatically

For the quickest repair of small, individual audio artifacts like isolated clicks or pops, use the Spot Healing Brush 



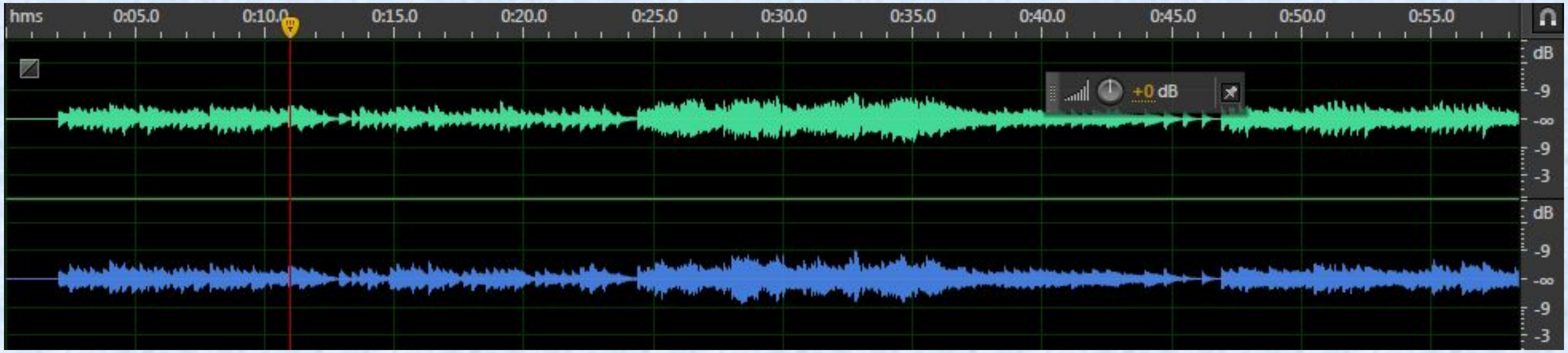
Example:
coughinmusic.wav

***A** Before*
***B** After*

Editing Audio

Adjust amplitude using amplitude control

By default, the visual amplitude control appears in a heads-up display (HUD) that floats over all waveforms

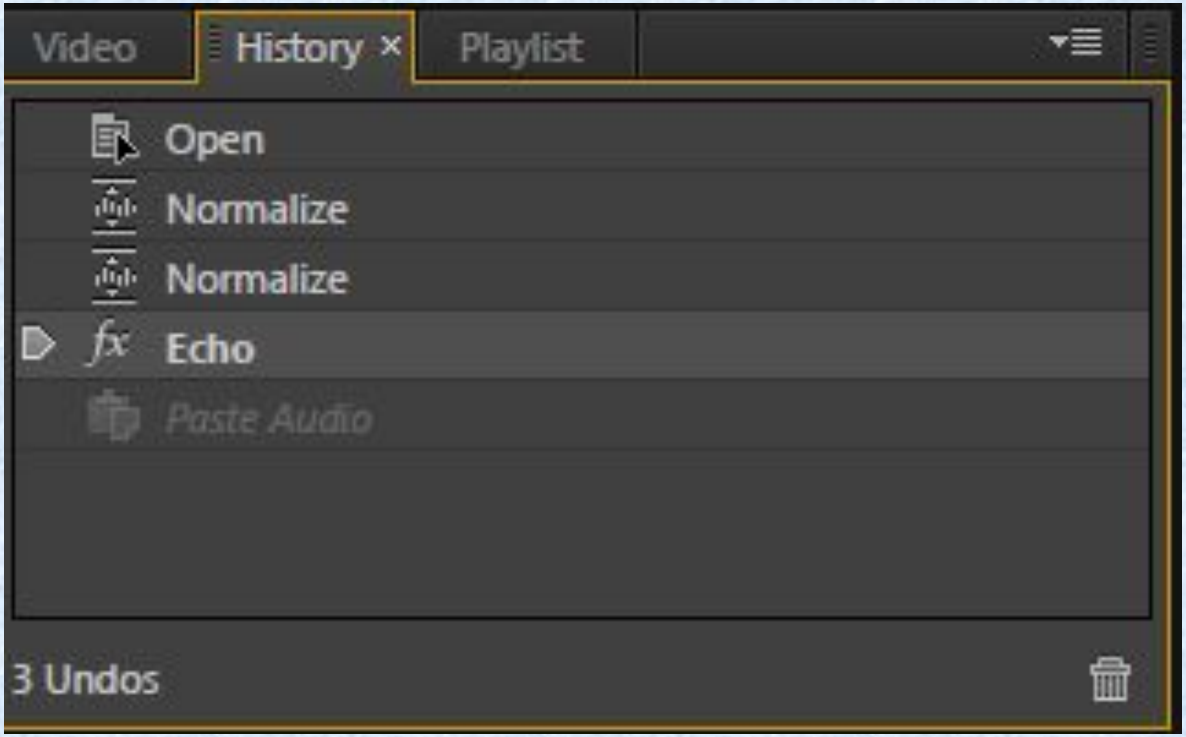


Example: tooquite.wav

Editing Audio

History Panel

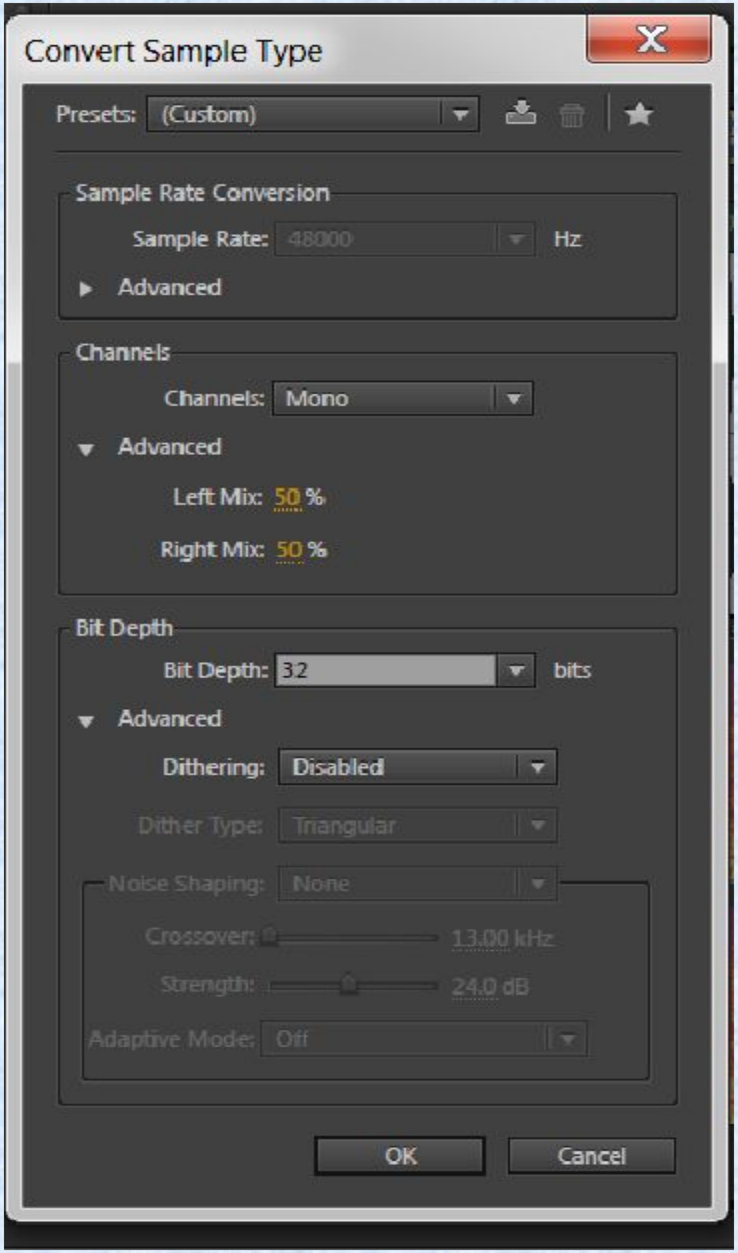
The History panel lets you instantly revert back to any previous change.



Editing Audio

EDIT>Converting sample types

- Convert the sample rate of a file
- Convert a waveform between surround, stereo, and mono
- Change the bit depth of a file



Exercise1 of Edit Audio

ex1File1: “I finished my work on Monday”

ex1File3: “I did finished my work on Saturday”

ex1File2: “I did not do my work on Saturday”

- **Steps**
  - Open ex1File1 and ex1File2
  - Find and Copy “did” in ex1File2
  - Paste on ex1File1
  - Delete “Monday” in ex1File1
  - Find and Copy “Saturday” in ex1File2
  - Paste on ex1File1
  - Save as ex1File3

Editing Audio

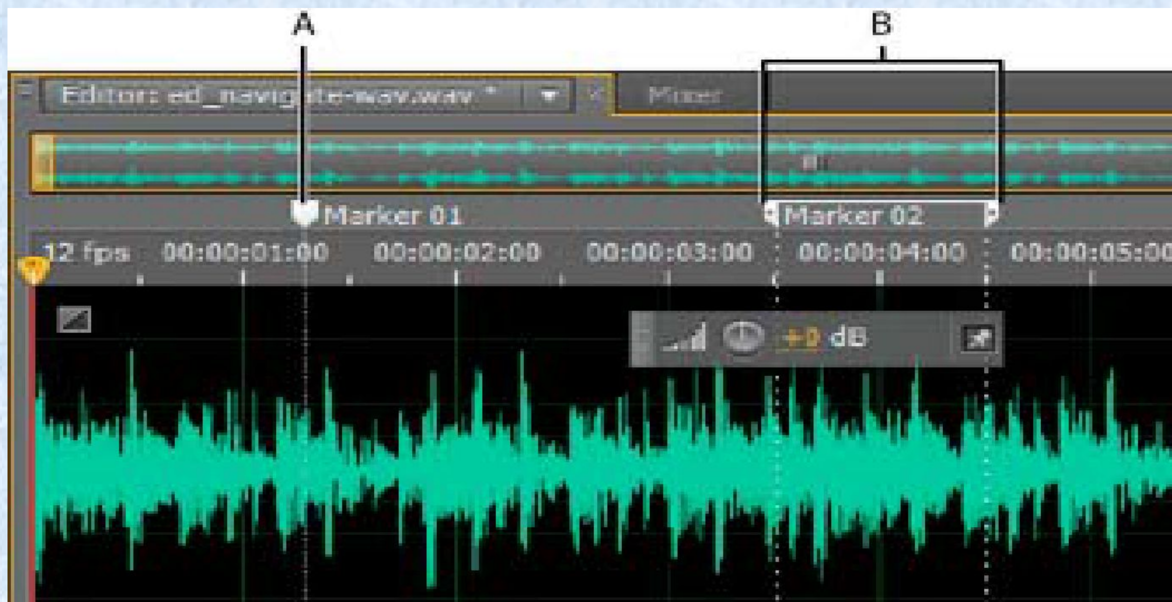
Working with markers using mark panel

Markers (sometimes called *cues*) are locations that you define in a waveform. A marker can be either a *point* or a *range*

- Either press the M key, or click the Add Marker button  in the Markers panel.
- Double-click a marker in the Markers panel to move the current-time indicator to that marker and select the area for range markers.

A Marker point

B Marker range



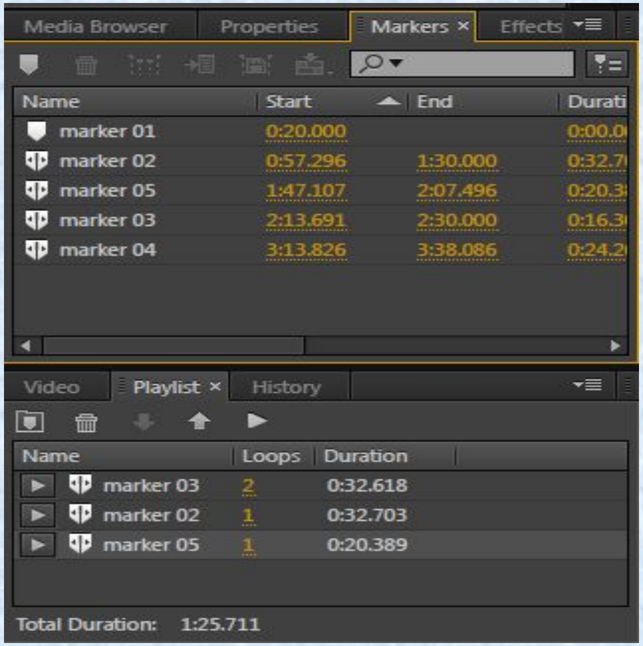
Editing Audio

Playlists

A *playlist* is an arrangement of marker ranges that you can play back in any order. A playlist lets you try different versions of an arrangement before you commit to edits.

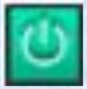
Create a playlist

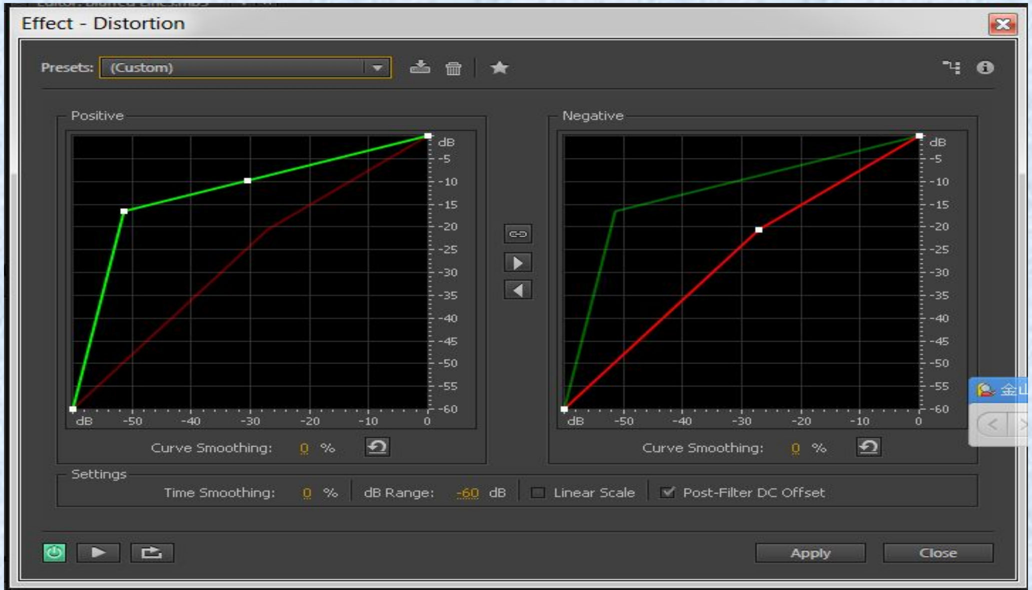
- Drag the range markers to the Playlist panel



Applying Effects

Apply individual effects

- From any submenu in the Effects menu, choose an effect.
- Click the Preview button , and then edit settings as needed.
- To compare original audio to processed audio, select and deselect the Power button .
- To apply the changes to the audio data, click Apply.

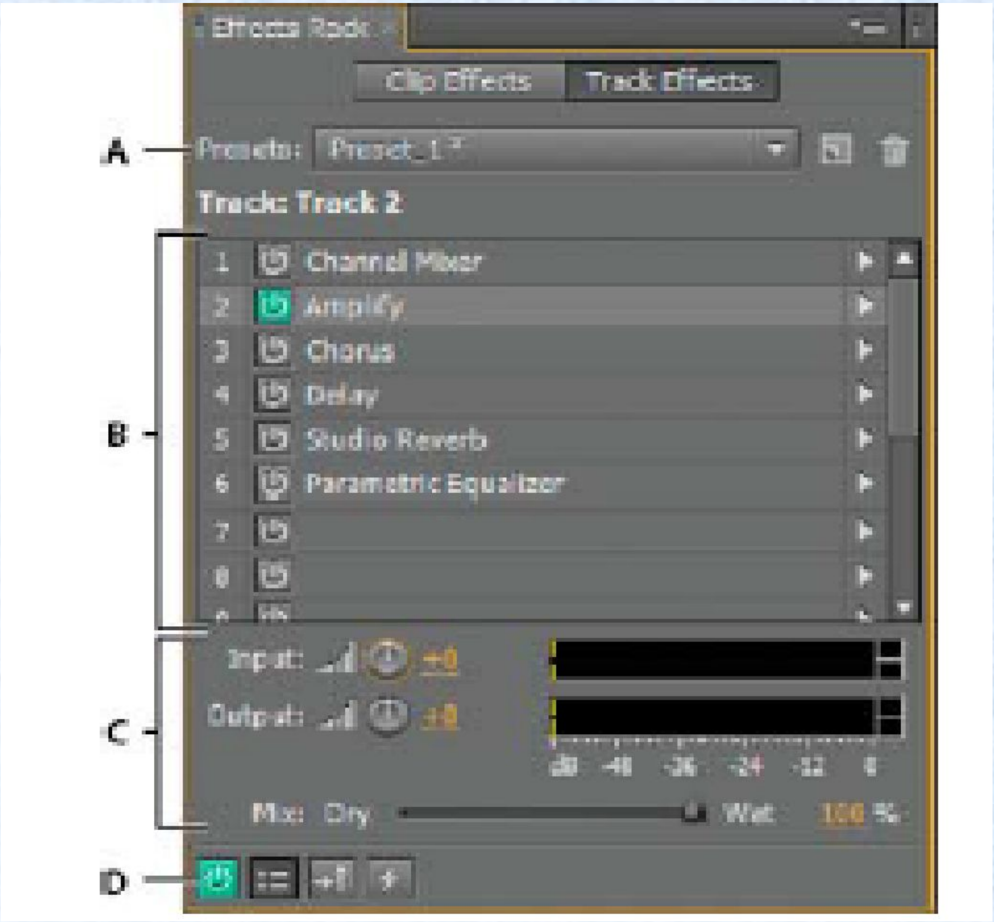


Example:
Convolution reverb

Applying Effects

Apply groups of effects

- The Effects Rack lets you insert, edit, and reorder up to 16 effects, optimize input, output and mix levels.



***A** Rack Preset controls*
***B** Effect slots*
***C** Level controls*
***D** Main Power button*

Applying Effects

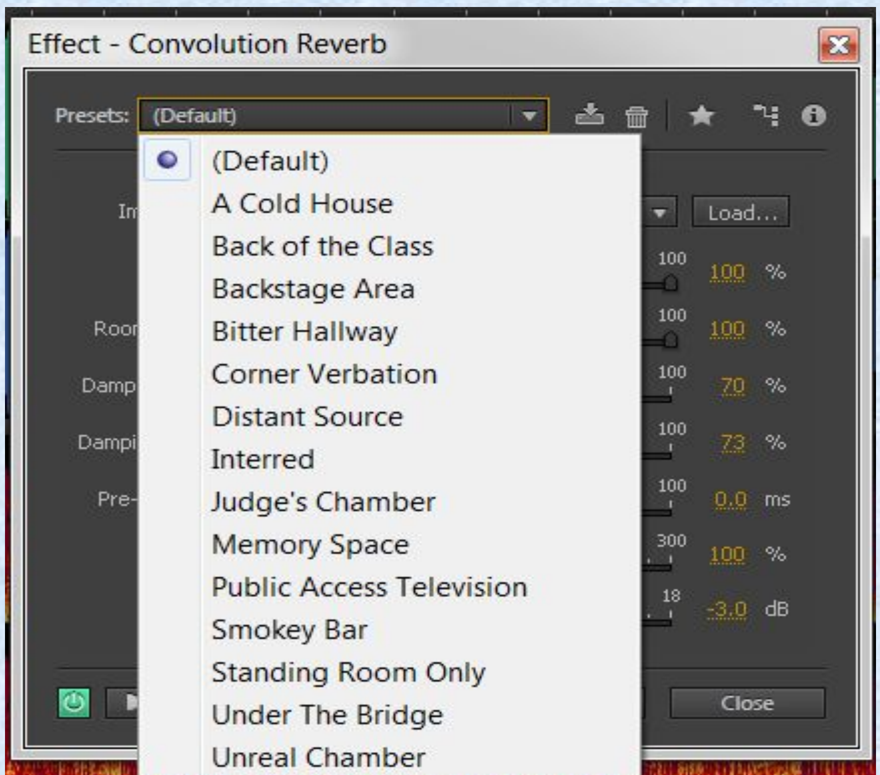
About process effects

- These processing-intensive effects can be applied only individually, so they aren’t accessible in the Effects Rack.

Applying Effects

Use effect presets

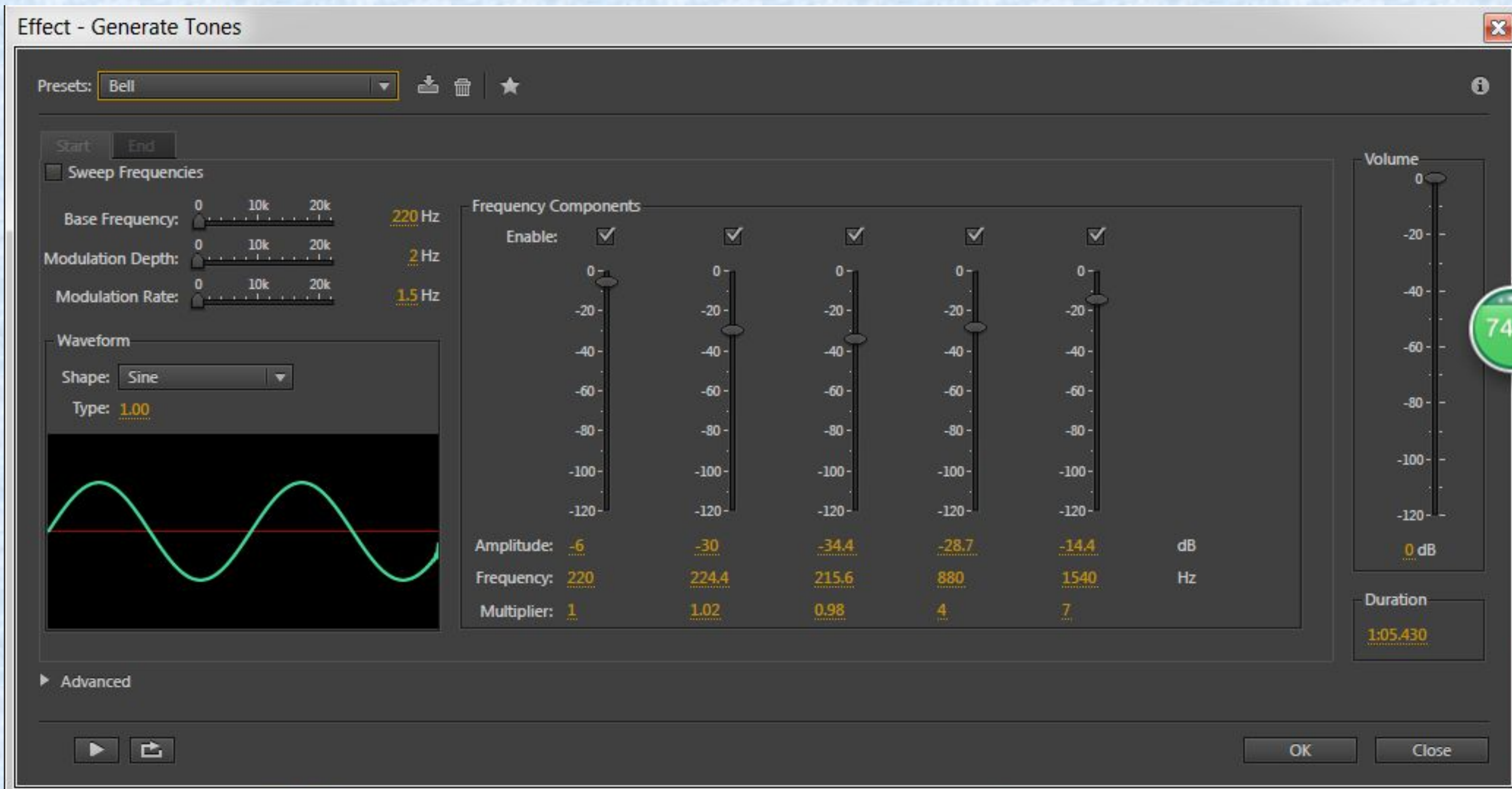
- Many effects provide presets that let you store and recall favorite settings. In addition to effect-specific presets, the Effects Rack provides rack presets that store groups of effects and settings.



Applying Effects

Generate a simple waveform

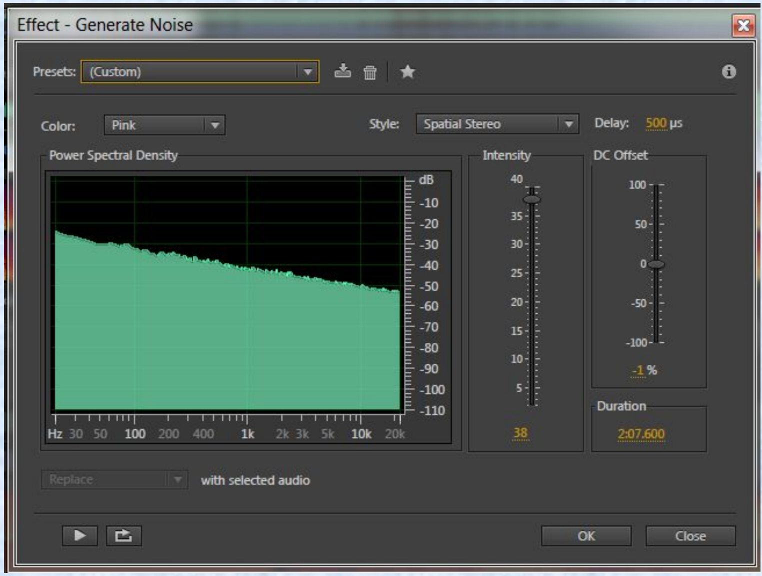
- Choose Effects > Generate > Tones to create a simple waveform using several amplitude- and frequency-related settings.



Applying Effects

Generate Noise

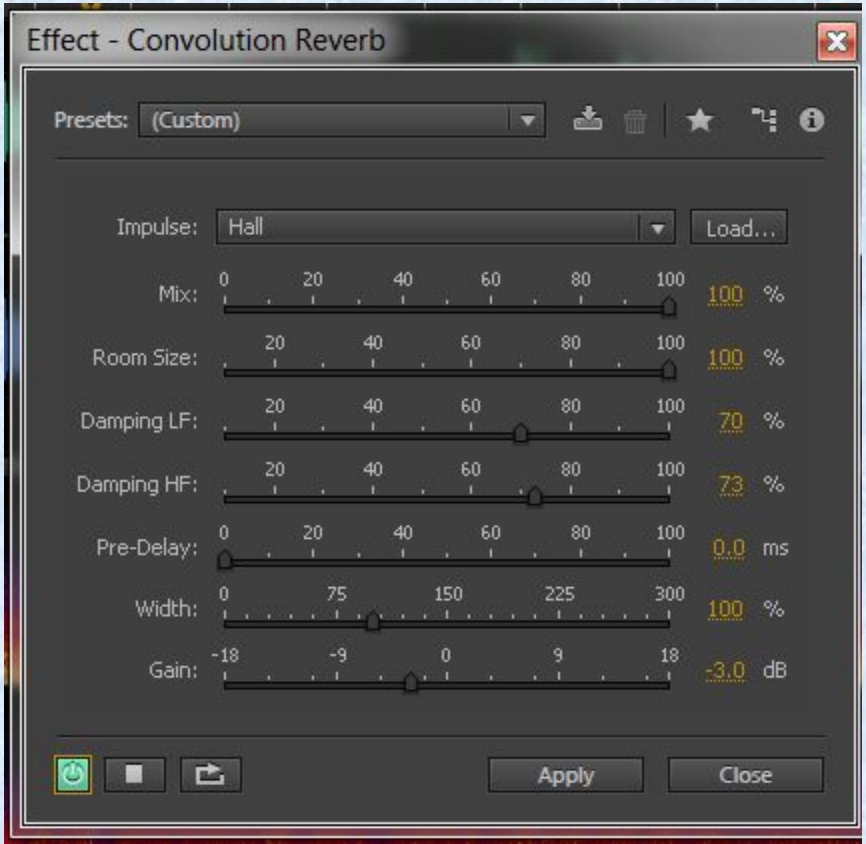
- Generating noise is useful for creating soothing sounds like waterfalls and for generating signals that can be used to check out the frequency response of a speaker, microphone, or other audiosystem component.
  - Place the cursor where you want to insert the noise. Or, if you want to replace part of the existing waveform, select the desired range of audio data.
  - Choose Effects > Generate > Noise.



Applying Effects

Reverb effects

In a room, sound bounces off the walls, ceiling, and floor on the way to your ears as a sonic surrounding that creates an impression of space. This reflected sound is called *reverb.* Reverb effects can be used to simulate a variety of room environments.



Impulse Specifies a file that simulates an acoustic space.

Applying Effects

Background Noise Reduction

The Noise Reduction effect dramatically reduces background and broadband noise with a minimal reduction in signal quality. This effect can remove a combination of noise, including tape hiss, microphone background noise, power-line hum, or any noise that is constant throughout a waveform.

- In the Waveform Editor, select a range that contains only noise and is at least half a second long.
- Effects > Noise Reduction/Restoration > Capture Noise Print.
- In the Editor panel, select the range from which you want to remove noise.
- Choose Effects > Noise Reduction/Restoration > Noise Reduction.

Exercise3 of Noise Reduction

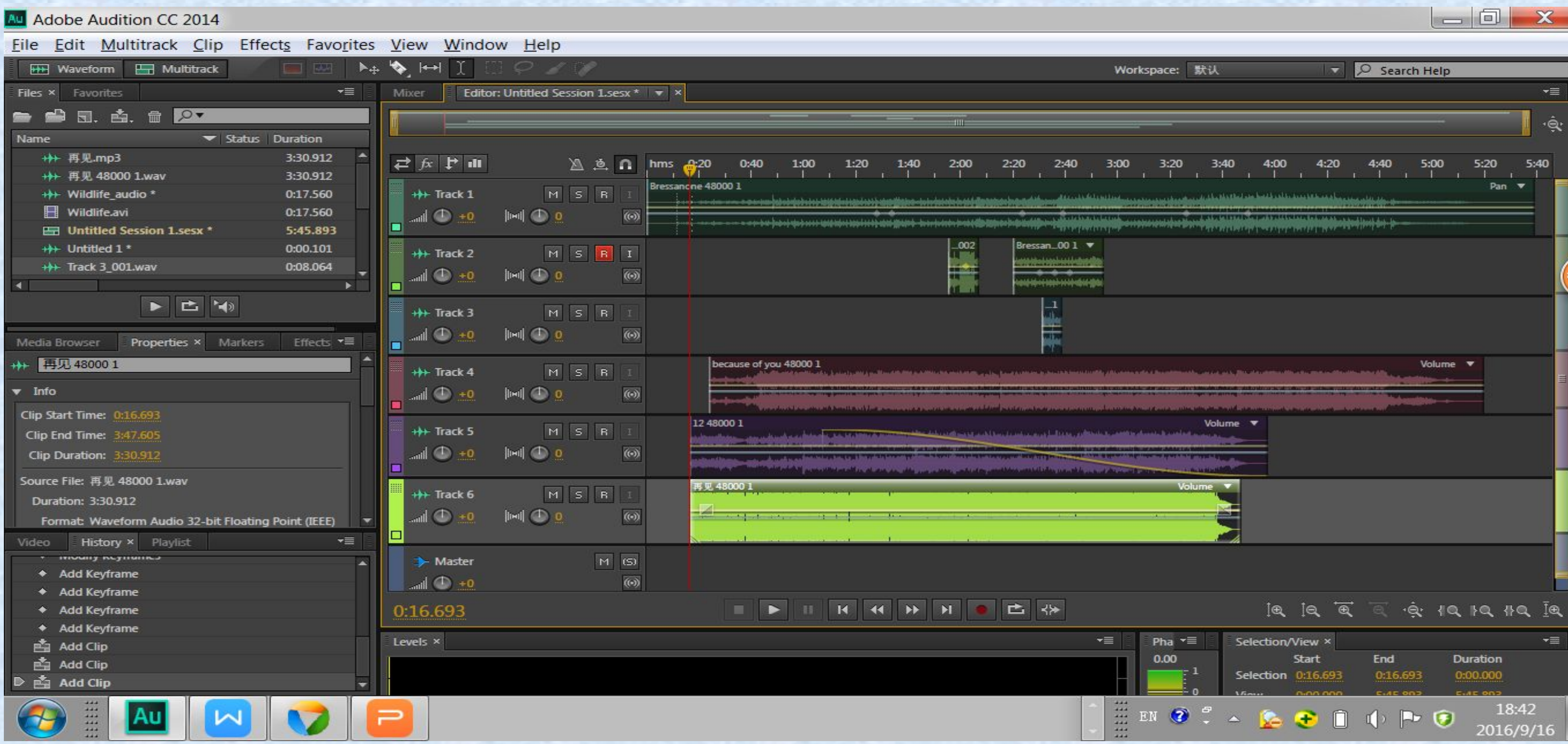
Ex2.wav

"I finished my work on Monday" + *Noise* → "I finished my work on Monday"

- **Steps**
  - Create audio file
  - Record "I finished my work on Monday"
  - Mix with noise
  - Open ex2.wav
  - Select a range that contains only noise and is at least half a second long.
  - Effects > Noise Reduction/Restoration > Capture Noise Print.
  - Select the range from which you want to remove noise.
  - Choose Effects > Noise Reduction/Restoration > Noise Reduction

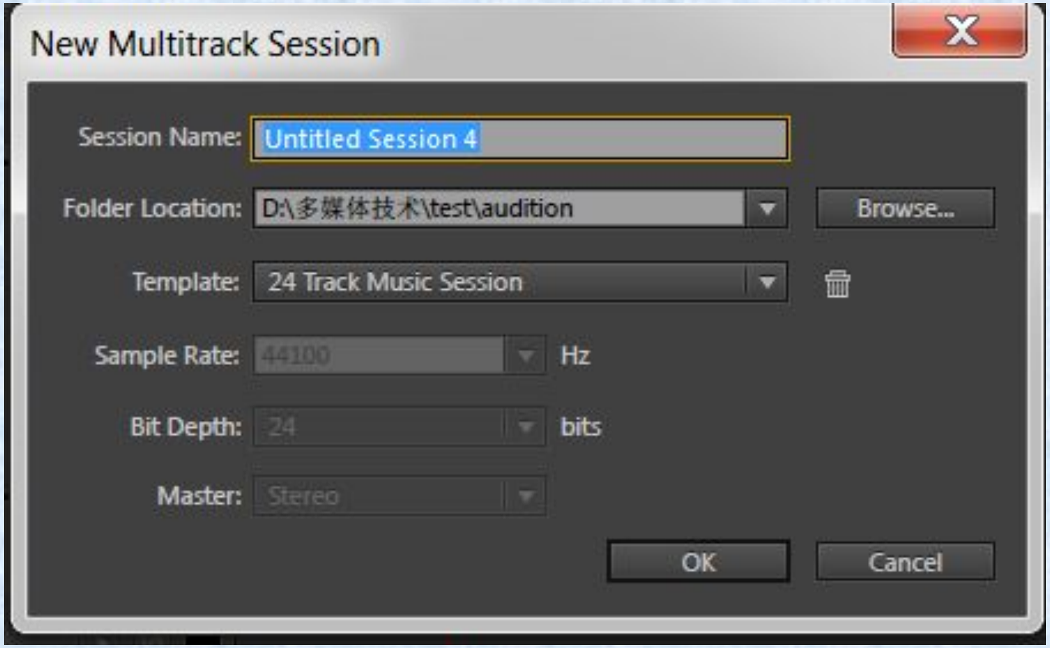
Mixing multitrack sessions

The Multitrack Editor can mix together multiple audio tracks to create layered soundtracks and elaborate musical compositions. You can record and mix unlimited tracks, and each track can contain as many clips as you.



Mixing multitrack sessions

- **Create a new multitrack session**
  - **Template:** specify source files and settings such as Sample Rate and Bit Depth.
  - **Sample Rate:** must shared by all files added to a session
  - **Bit Depth:** cannot be changed after a session is created



Mixing multitrack sessions

- **Insert an audio file into a track**
  - Place CTI at the desired time position in a track.
  - Choose Multitrack > Insert File.

  The inserted file becomes an audio clip on the selected track.
- **Record an audio clip on multiple tracks by overdubbing**
  - Click the Arm For Record buttons for the tracks,
  - Click the Record button to start and stop recording.

  Each recording becomes a new audio clip on a track.

Mixing multitrack sessions

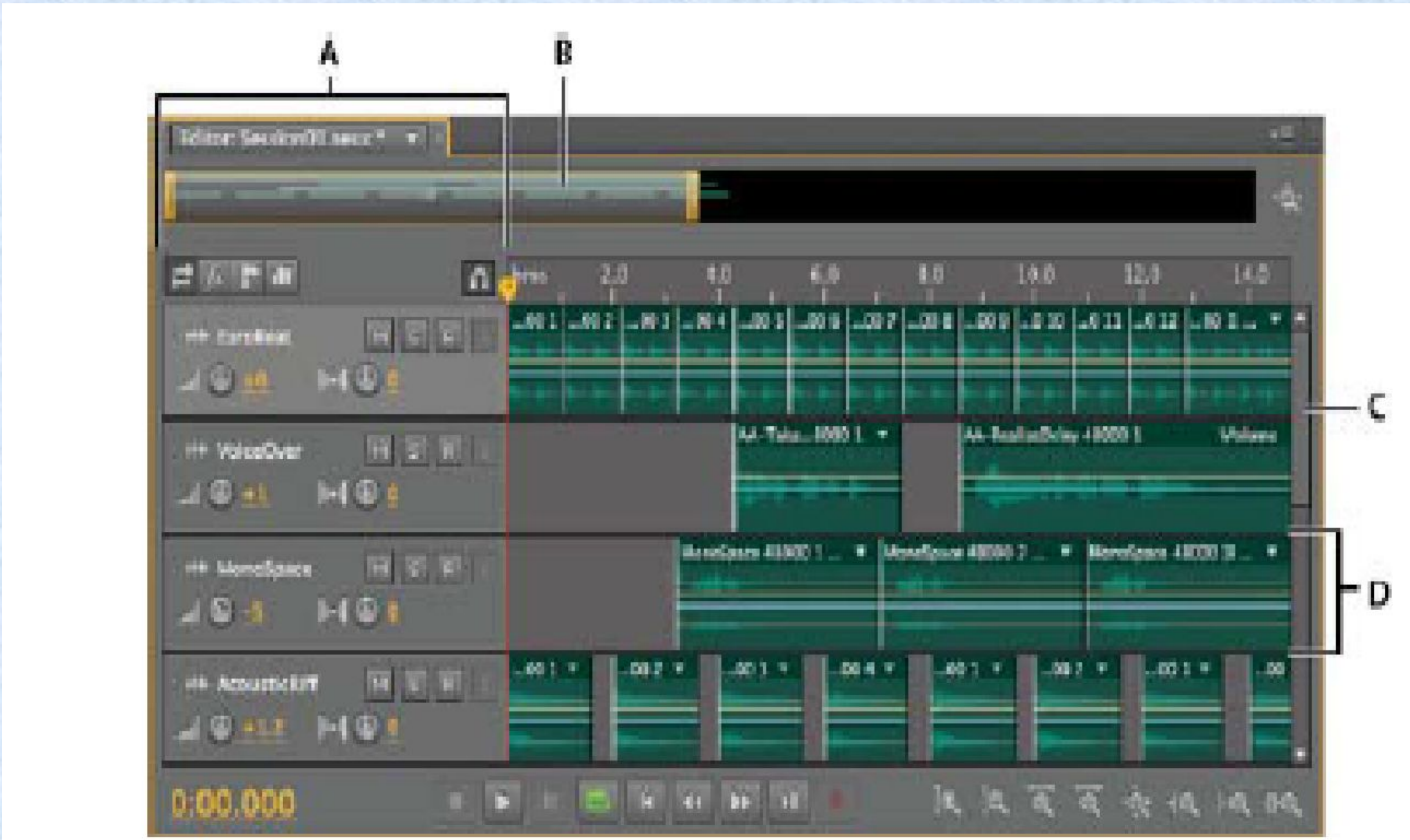
Session (*.sesx) files

- Adobe Audition save multitrack sessions in session (.sesx) files which contain no audio data themselves. A session file is a small XML-based file which keeps track of which files are a part of the session, where they are inserted, which envelopes and effects are applied, and so on.

<?xml version="1.0" encoding="UTF-8" standalone="no" ?><!DOCTYPE sesx><sesx version="1.0"> <session appVersion="7.0" audioChannelType="stereo" bitDepth="32" duration="1440000" sampleRate="48000"> <name>Untitled Session 6.sesx</name> <tracks> <audioTrack automationLaneOpenState="false" id="10001" index="1" select="true" visible="true"> <trackParameters trackHeight="134" trackHue="-1.00" trackMinimized="false"> <name>Track 1</name> </trackParameters> <trackAudioParameters audioChannelType="stereo" automationMode="1" monitoring="false" recordArmed="false" solo="false" soloSafe="false"> <trackOutput outputID="10000" t

Mixing multitrack sessions

Multitrack Editor



__A__ Track controls __B__ Zoom navigator __C__ Vertical scroll bar __D__ Track

Mixing multitrack sessions

Arranging and editing tracks

- **Add or delete tracks**
- **Name tracks**
- **Move tracks**
- **Mute and solo tracks**
- **Set track output volume**
- **….**

Editing multitrack clips

- **Move a clip**
- **Copy a clip**
- **Remove a selected range from clips**
- **Trimming and extending clips**
    - position the cursor over the left or right edge of the clip and drag clip edges
- **Split clips**
- **….**

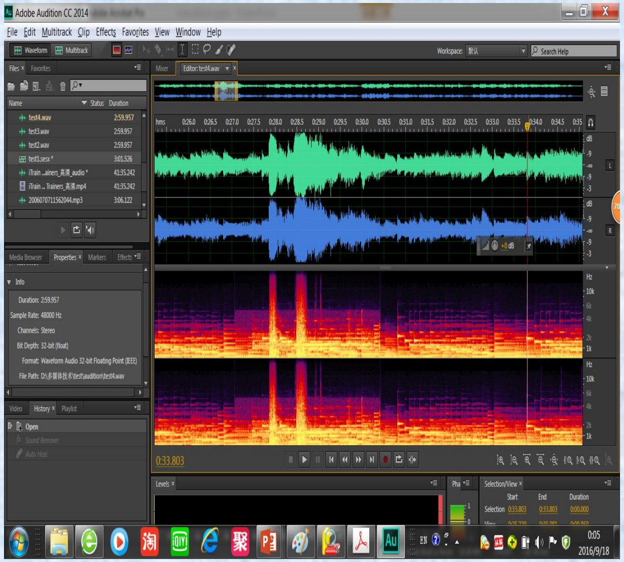
Mixing multitrack sessions

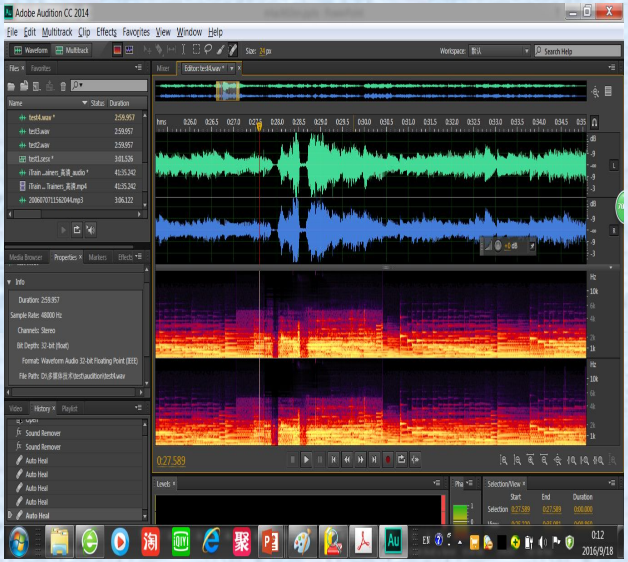
Export multitrack mixdown files

After you finish mixing a session, you can export all or part of it in a variety of common formats.

- **Choose File > Export > Multitrack Mixdown.**

Exercise 4 of Sound Remove

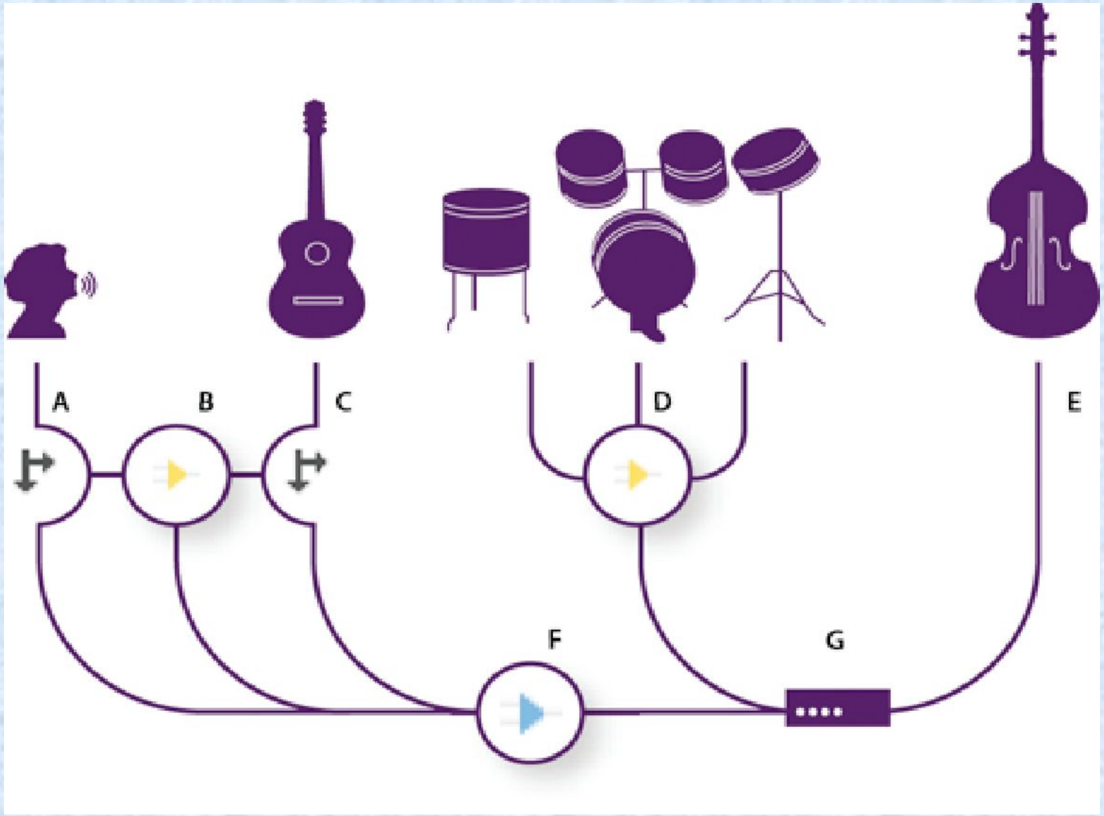




- **Steps**
  - Mix cough with music in multitrack
  - Remove cough using Sound Remove effect
  - Refine your work using Spot Healing Bush tool

Multitrack routing

Buses, sends, and the Master track let you route multiple track outputs to one set of controls. With these combined controls, you can efficiently organize and mix a session.



Mixing multitrack sessions

Audio tracks 

Audio tracks contain either imported audio or clips recorded in the current session. These tracks offer the widest range of controls, letting you specify inputs and outputs, apply effects and equalization, route audio to sends and buses, and automate mixes.

Mixing multitrack sessions

Assign audio inputs and outputs to tracks

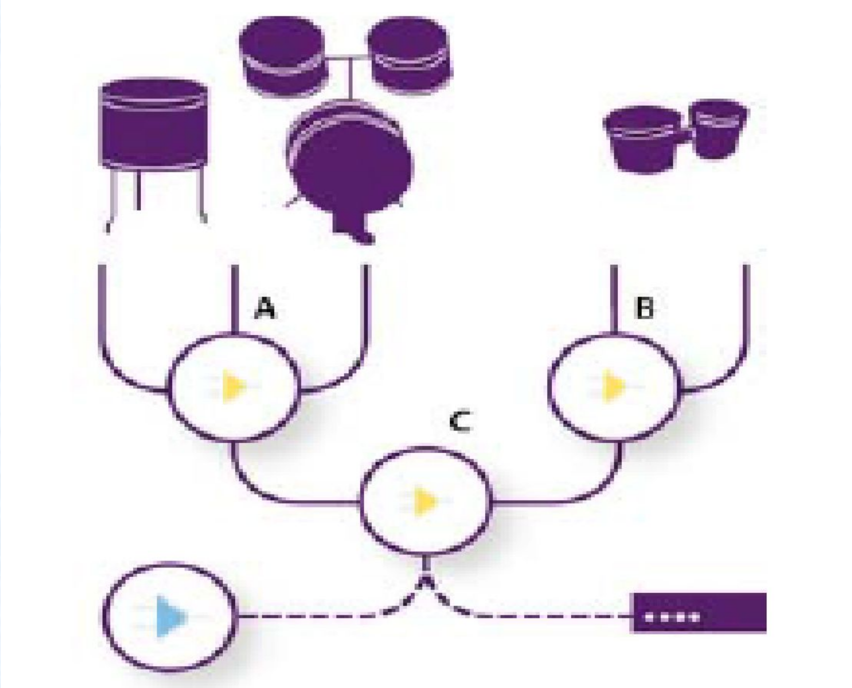
In the Inputs/Outputs area  of the Editor panel, do the following:

- From the Input menu, choose a hardware input.
- From the Output menu, choose a bus, the Master track, or a hardware output.

Mixing multitrack sessions

Bus tracks

With bus tracks, you can combine the outputs of several audio tracks or sends and control them collectively. For example, to control the volume of multiple drum tracks with a single fader, or, to optimize system performance, apply a single reverb effect to a bus track.

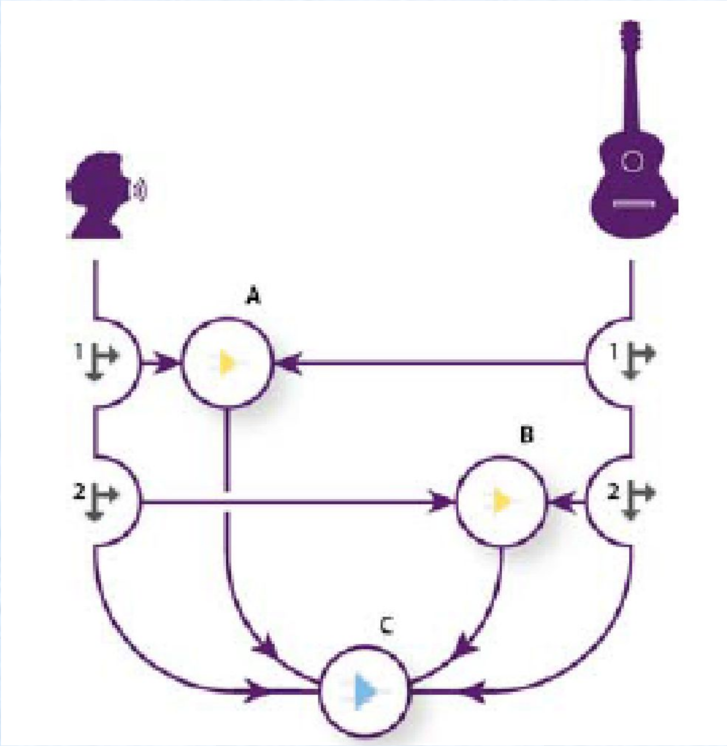


A *Drum kit bus*
B *Hand drum bus*
C *Combined drums bus outputting to either the Master track or hardware*

Mixing multitrack sessions

Sends

Sends let you route audio from a track to multiple buses, creating tremendous signal-routing flexibility. Each track provides up to 16 sends in the Send area , which you configure independently from the track output.



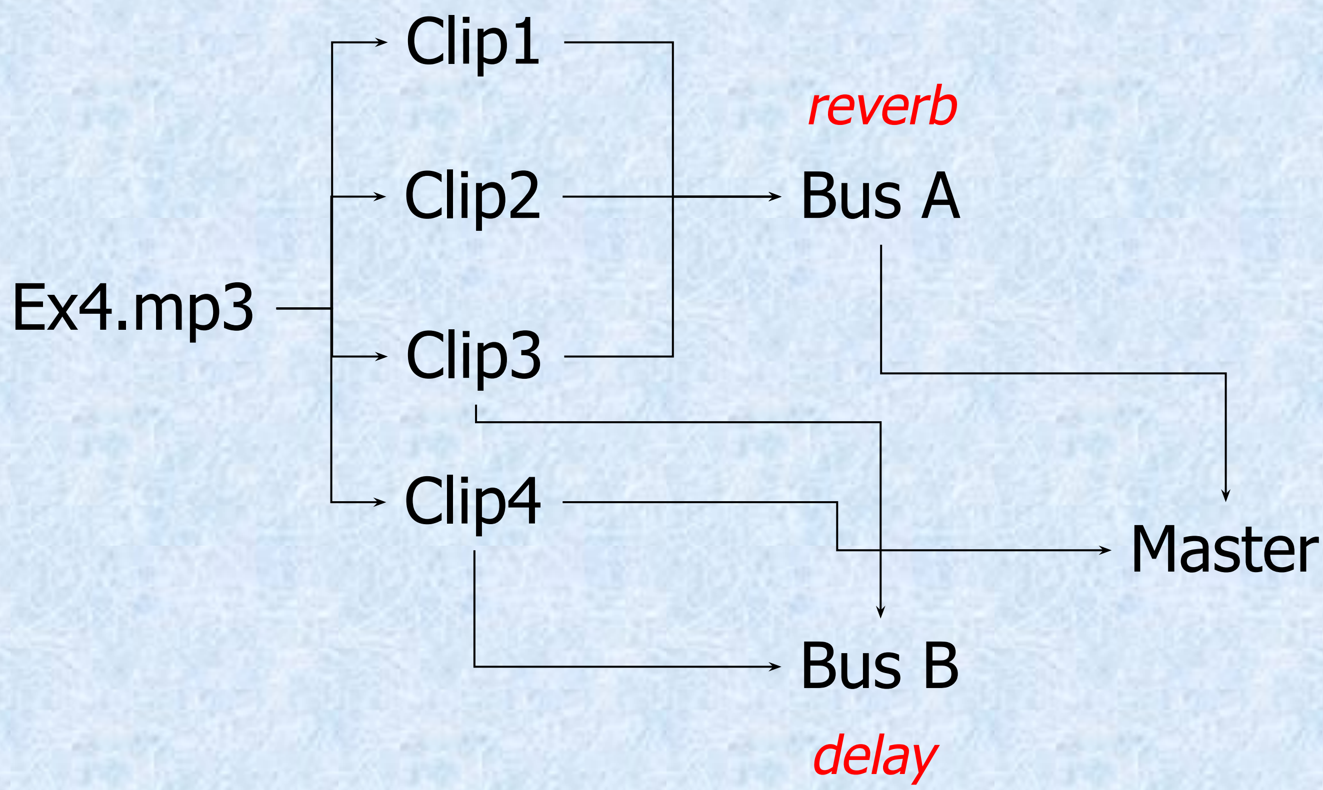
A *Send 1 outputs to delay bus*
B *Send 2 outputs to reverb bus*
C *Master track combines vocal, guitar, delay, and reverb outputs*

Mixing multitrack sessions

Master track 

The Master track , which is the last in each session, lets you easily combine the outputs of multiple tracks and buses and control them with a single fader. A session always contains one Master track. The Master track can’t directly connect to audio inputs, or output to sends or buses; it can only output directly to hardware ports.

Exercise 5 of Multitrack routing



Summary

☆**Audio Digitalization, Music Symbolization**

☆**Three Essentials of Digital Audio, Audio File Formats**

☆**Digital Audio Compression Standards and Software**

☆**Sound Card and Electroacoustic Equipment**

☆**Electric Music and MIDI**

☆**Audio Processing Software: Audition**