Audio

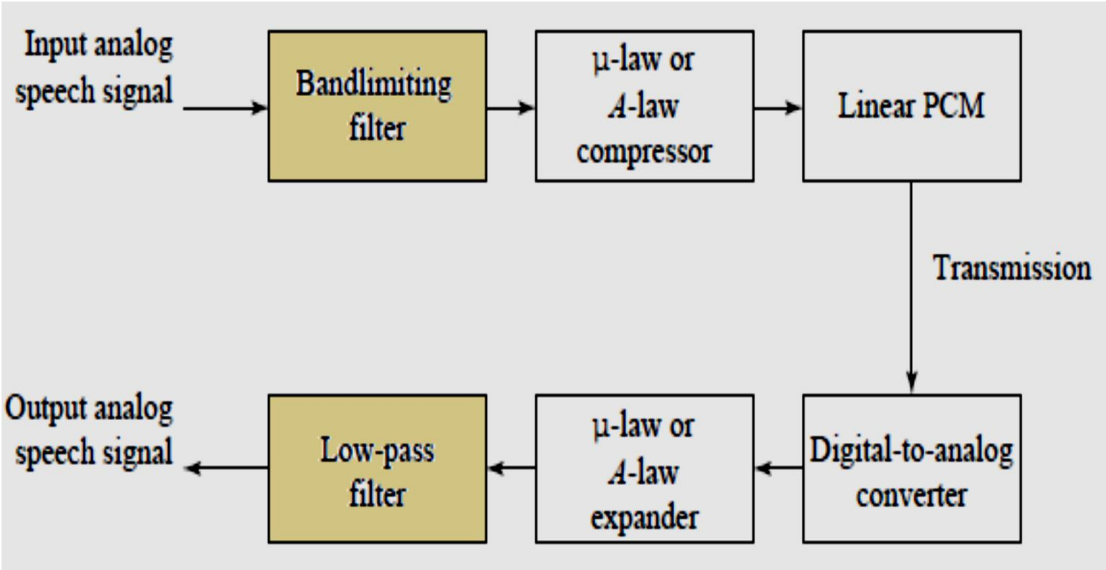
Audio Filtering

(a) Prior to sampling and AD conversion, the audio signal is usually filtered to remove unwanted frequencies by a ***band pass filter***. The frequencies kept depend on the application:

- For speech, typically from 50Hz to 10kHz is retained, and other frequencies are blocked by the use of a band-pass filter that screens out lower and higher frequencies.
- An audio music signal will typically contain from about 20Hz up to 20kHz.

(b) At the DA converter end, high frequencies may reappear in the output because of sampling and then quantization, smooth input signal is replaced by a series of step functions containing all possible frequencies. So at the decoder side, a lowpass filter is used after the DA circuit.

The sampled signal can be returned to the continuous-time domain simply by passing it into a ***low-pass filter***. This filter can be called a reconstruction filter.



Synthetic Sounds

- Can we create sound wave rather than record and play it back ??

In fact, we can create it on the computer and skip the recording. We call this process *sound synthesis*. A synthesizer was a stand alone sound generator that can vary pitch, loudness, and tone. Synthesizer: electronic musical instrument operated by keyboard, producing a wide variety of sounds by generating and combining signals of different frequencies

Synthesizers are computer programs that create sound waves from scratch, it can shaping sound waves — for taking one wave and twisting it into another. Music synthesis can produce almost limitless sounds, including many that we can’t find in nature.

There are two fundamentally different approaches to handling stored sampled audio.

1. Frequency Modulation FM synthesis: more interesting sound is created by changing the argument of the main sinusoid term.

2. Wave Table synthesis: A more accurate way of generating sounds from digital signals. In this technique, the actual digital samples of sounds from real instruments are stored.

Since wave tables are stored in memory on the sound card, they can be manipulated by software so that sounds can be combined, edited, and enhanced. Wave table synthesis is more accurate and expensive than FM synthesis, partly because the data storage needed is much larger.

MIDI: Musical Instrument Digital Interface

MIDI forms a protocol adopted by the electronic music industry that enables computers, synthesizers, keyboards, and other musical devices to communicate with each other.

It is a scripting language that codes "events or messages" that stand for the production of sounds. E.g., a MIDI event might include values for the pitch of a single note, its duration, and its volume.

A *MIDI keyboard* produces no sound, instead generating sequences of MID instructions (messages), and since MIDI files don't contain sampled audio like MP3 or WAV files, they're comparatively much smaller than audio files. A minute of compressed audio adds up to around 10Mb of data, while a minute of sound translated into MIDI only takes up 10Kb. This makes MIDI a great choice for memory-starved devices like cell phones and video games.

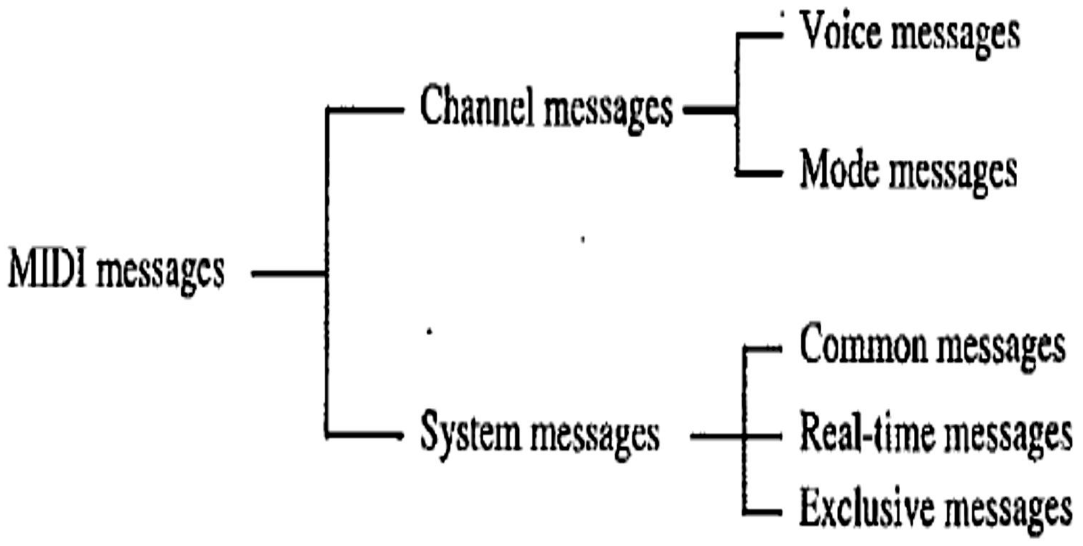
MIDI channels are used to separate messages. There are 16 channels, numbered from 0 to 15. The idea is that each channel is associated with a particular instrument – for example, channel 1 is the piano, channel 10 is the drums.

When you record music onto a computer using MIDI, the software saves this list of messages and instructions as a .MID file. If you play the .MID file back on an electronic keyboard, the keyboard's internal synthesizer software follows the instructions to play back the song. The keyboard will play a certain key with a certain velocity and hold it for a specified amount of time before moving on to the next note.

MIDI messages can be classified into two types:

1- Channel messages: special message to an instrument's channel.

2- system messages: general message for all instruments indicating a change in tuning or timing such as timing signals for synchronization, positioning information in pre-recorded, MIDI sequences, and detailed setup information for the destination device..



Here are a few examples of typical MIDI messages:

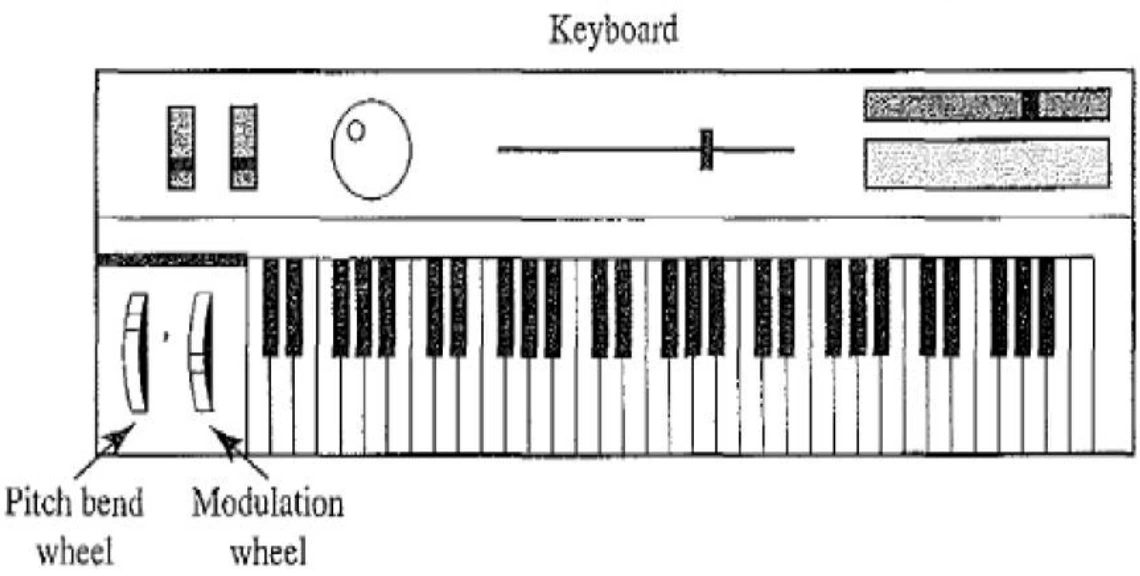
1- Note On: signals that a key has been pressed or a note on another instrument (like a MIDI guitar or clarinet) has been played. The Note On message includes instructions for what key was pressed, what channel, what pitch, and at what volume.

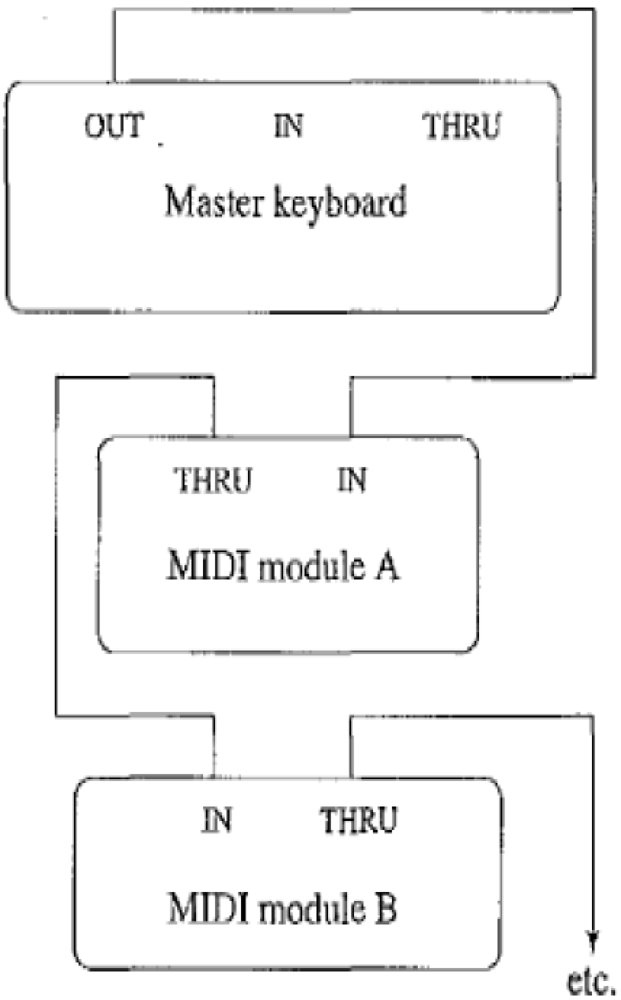
2- Note Off: signals that the key has been released or the note is done playing.

3- Polyphonic Key Pressure is a measurement of how hard a key is pressed. On some keyboards, this adds vibrato or other effects to the note.

4- Control Change indicates that a controller has been pressed or turned.

5- Pitch Wheel Change signals.


Keyboard
Pitch bend wheel
Modulation wheel


OUT
IN
THRU
Master keyboard
THRU
IN
MIDI module A
IN
THRU
MIDI module B
etc.

Quantization and Transmission of Audio

Quantization and transformation of data are collectively known as coding of the data. For audio, the μ-law technique for companding audio signals is usually combined with an algorithm that exploits the temporal redundancy present in audio signals. Where companding algorithms in digital domain reduce the quantization error (hence increasing signal to quantization noise ratio).

Differences in signals between the present and a past time can reduce the size of signal values and also concentrate the histogram of pixel values into a much smaller range.

The result of reducing the variance of values is that lossless compression methods produce a bitstream with shorter bit lengths for more likely values.

In general, producing quantized sampled output for audio is called

1- PCM (Pulse Code Modulation).

3- The differences version is called **DPCM**

4- crude but efficient variant is called **DM**

5- The adaptive version is called **ADPCM**.

Pulse Code Modulation PCM

_ The basic techniques for creating digital signals from analog signals are sampling and quantization.
Sampling is invariably done uniformly – we select a sampling rate and produce one value for each sampling time.

In the magnitude direction, we digitize by quantization, selecting breakpoints in magnitude and remapping any value within an interval to one representative output level.
Assuming a bandwidth for speech from about 50 Hz to about 10 kHz, the Nyquist rate would dictate a sampling rate of20 kHz. Using uniform quantization without companding, the minimum sample size we could get away with would likely be about 12 bits. Hence, for mono speech transmission the bitrate would be 240 kbps. With companding, we can

safely reduce the sample size to 8 bits with the same perceived level of quality and thus reduce the bitrate to 160 kbps. However, the standard approach to telephony assumes that the highest-frequency audio signal we want to reproduce is about 4 kHz. Therefore, the sampling rate is only 8 kHz, and the companded bitrate thus reduces to only 64 kbps.

Differential Coding of Audio

Audio is often stored not in simple PCM but in a form that exploits differences. For a start, differences will generally be smaller numbers and hence offer the possibility of using fewer bits to store.

Streaming Audio

- Streaming audio means sound is delivering over a network and played as it arrives without having to be stored on the computer.
- Because of lower bandwidth required by audio, streaming is more successful for sound than it is for video.

- Example for streaming audio format: Real Network, RealAudio, streaming QuickTime.
- Software required for playing streaming audio: RealPlayer, and Windows media Player.