

# Compression:

## Audio Compression (MPEG and others)

As with video a number of compression techniques have been applied to audio.

### Simple Audio Compression Methods

#### RECAP (Already Studied)

Traditional lossless compression methods (Huffman, LZW, etc.) usually don't work well on audio compression

- For the same reason as in image and video compression:

**Too much change variation in data over a short time**



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# Some Simple But Limited Practical Methods

- Silence Compression - detect the "silence", similar to run-length encoding (**seen examples before**)

- Differential Pulse Code Modulation (DPCM)

Relies on the fact that difference in amplitude in successive samples is small then we can use reduced bits to store the difference (**seen examples before**)



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## Simple But Limited Practical Methods (Cont.)

- Adaptive Differential Pulse Code Modulation (ADPCM)  
e.g., in CCITT G.721 – 16 or 32 Kbits/sec.
  - (a) Encodes the difference between two consecutive signals but a refinement on DPCM,
  - (b) Adapts at quantisation so fewer bits are used when the value is smaller.
    - It is necessary to predict where the waveform is heading  
→ **difficult**
    - Apple had a proprietary scheme called ACE/MACE. Lossy scheme that tries to predict where wave will go in next sample. About 2:1 compression.

## Simple But Limited Practical Methods (Cont.)

- Adaptive Predictive Coding (APC) typically used on Speech.
  - Input signal is divided into fixed segments (**windows**)
  - For each segment, some sample **characteristics** are computed, **e.g. pitch, period, loudness**.
  - These characteristics are used to predict the signal
  - Computerised talking (Speech Synthesisers use such methods) but low bandwidth:

**Acceptable quality at 8 kbits/sec**



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## Simple But Limited Practical Methods (Cont.)

- Linear Predictive Coding (LPC) fits signal to speech model and then transmits parameters of model as in APC.

Speech Model:

– Speech Model:

Pitch, period, loudness, vocal tract parameters (voiced and unvoiced sounds).

- Synthesised speech
- Still sounds like a computer talking,
- Bandwidth as low as 2.4 kbits/sec.

## Simple But Limited Practical Methods (Cont.)

- Code Excited Linear Predictor (CELP) does LPC, but also transmits error term.
  - Based on more sophisticated model of vocal tract than LPC
  - Better perceived speech quality
  - Audio conferencing quality at 4.8 kbits/sec.



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# Psychoacoustics or Perceptual Coding

**Basic Idea:** Exploit areas where the human ear is **less sensitive** to sound to achieve compression

**E.g.** MPEG audio, Dolby AC.

How do we hear sound?

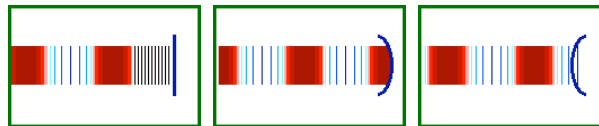


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# Sound Revisited

- Sound is produced by a vibrating source.
- The vibrations disturb air molecules
- Produce variations in air pressure: lower than average pressure, **rarefactions**, and higher than average, **compressions**. **This produces sound waves.**
- When a sound wave impinges on a surface (e.g. eardrum or microphone) it causes the surface to vibrate in sympathy:



- In this way **acoustic energy** is transferred from a source to a receptor.



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# Human Hearing

- Upon receiving the the waveform the eardrum vibrates in sympathy
- Through a variety of mechanisms the acoustic energy is transferred to nerve impulses that the brain interprets as sound.

The ear can be regarded as being made up of 3 parts:

- The outer ear,
- The middle ear,
- The inner ear.



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# Human Ear

We consider:

- The function of the main parts of the ear
- How the transmission of sound is processed.

⇒ FLASH EAR DEMO (Lecture ONLY)

[Click Here to Run Flash Ear Demo over the Web](#)

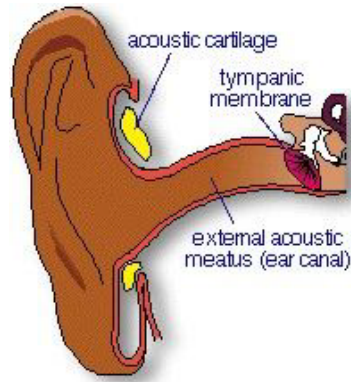
(Shockwave Required)



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# The Outer Ear



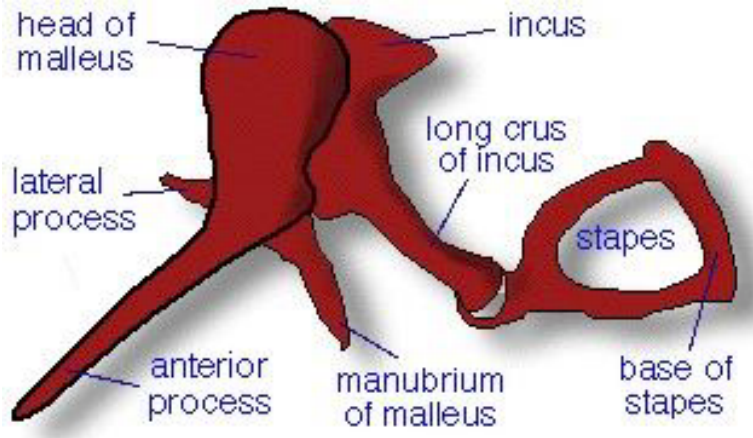
- **Ear Canal:** Focuses the incoming audio.
- **Eardrum (Tympanic Membrane):**
  - Interface between the external and middle ear.
  - Sound is converted into mechanical vibrations via the middle ear.
  - Sympathetic vibrations on the membrane of the eardrum.



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# The Middle Ear



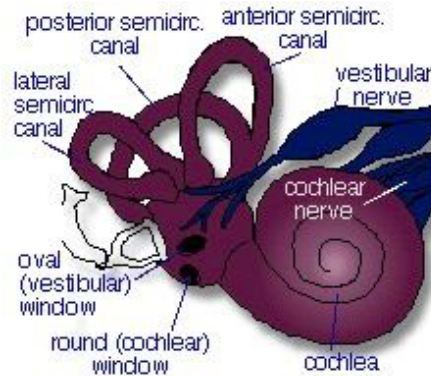
- 3 small bones, the **ossicles**:  
**Malleus**, **Incus**, and **Stapes**.
- Form a system of levers which are linked together and driven by the eardrum
- Bones amplify the force of sound vibrations.



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# The Inner Ear



## The Cochlea:

- Transforms mechanical ossicle forces into hydraulic pressure,
- The cochlea is filled with fluid.
- Hydraulic pressure imparts movement to the cochlear duct and to the organ of Corti.
- Cochlea which is no bigger than the tip of a little finger!!

## Semicircular canals

- Body's balance mechanism
- Thought that it plays no part in hearing.



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# How the Cochlea Works

- Pressure waves in the cochlea exert energy along a route that begins at the oval window and ends abruptly at the membrane-covered round window
- Pressure applied to the oval window is transmitted to all parts of the cochlea.

## Stereocilia

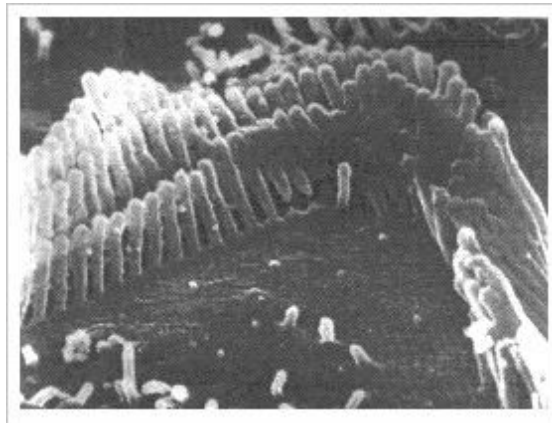
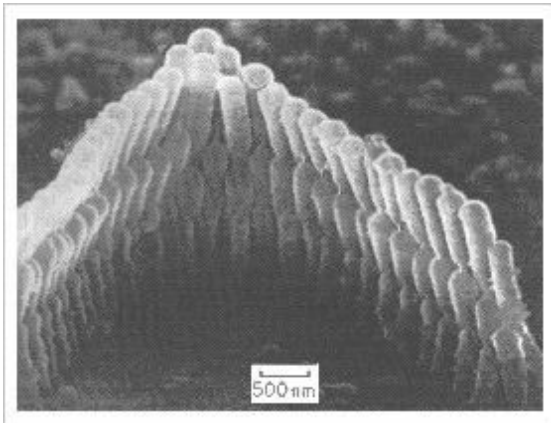
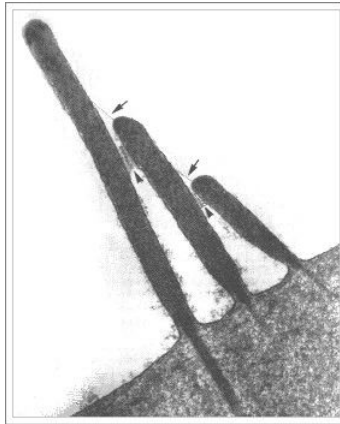
- Inner surface of the cochlea (**the basilar membrane**) is lined with over 20,000 hair-like nerve cells — **stereocilia**,
- One of the most critical aspects of hearing.



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# Stereocilia Microscope Images



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# Hearing different frequencies

- Basilar membrane is tight at one end, looser at the other
- High tones create their greatest crests where the membrane is tight,
- Low tones where the wall is slack.
- Causes resonant frequencies much like what happens in a tight string.
- Stereocilia differ in length by minuscule amounts
- they also have different degrees of resiliency to the fluid which passes over them.



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# Finally to nerve signals

- Compressional wave moves middle ear through to the cochlea
- Stereocilia will be set in motion.
- Each stereocilia sensitive to a particular frequency.
- Stereocilia cell will resonate with a larger amplitude of vibration.
- Increased vibrational amplitude induces the cell to release an electrical impulse which passes along the auditory nerve towards the brain.

In a process which is not clearly understood, the brain is capable of interpreting the qualities of the sound upon reception of these electric nerve impulses.



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# Sensitivity of the Ear

- Range is about 20 Hz to 20 kHz, most sensitive at 2 to 4 KHz.
- Dynamic range (quietest to loudest) is about 96 dB
- Approximate threshold of pain: 130 dB
- Hearing damage:  $> 90$  dB (prolonged exposure)
- Normal conversation: 60-70 dB
- Typical classroom background noise: 20-30 dB
- Normal voice range is about 500 Hz to 2 kHz
  - Low frequencies are vowels and bass
  - High frequencies are consonants

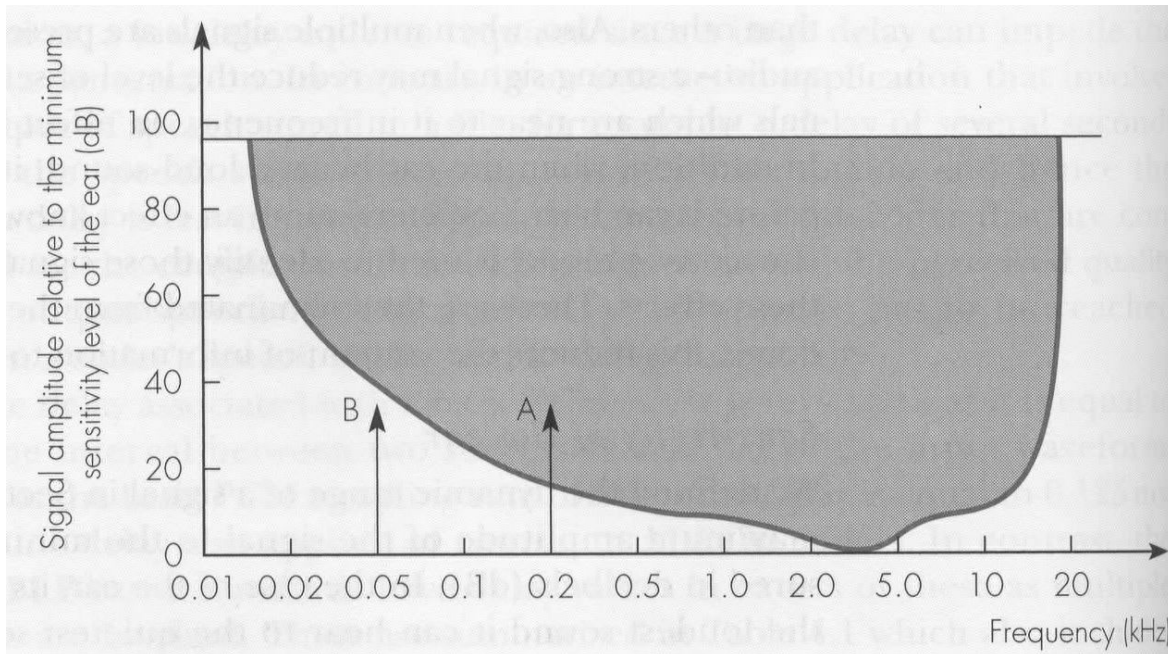


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# Question: How sensitive is human hearing?

The sensitivity of the human ear with respect to frequency is given by the following graph.



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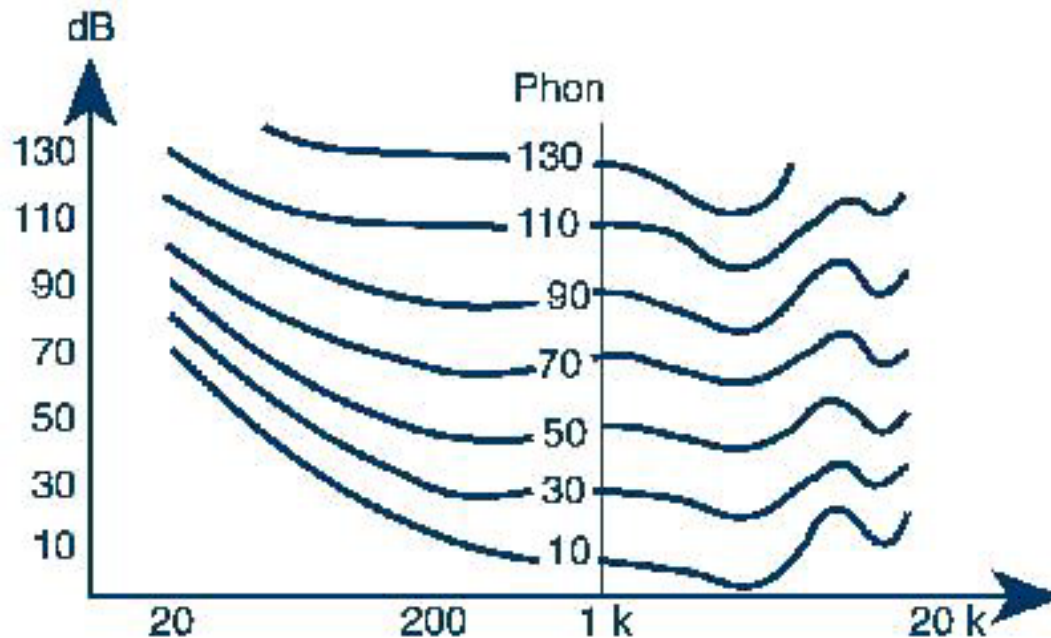
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## Frequency dependence is also level dependent!

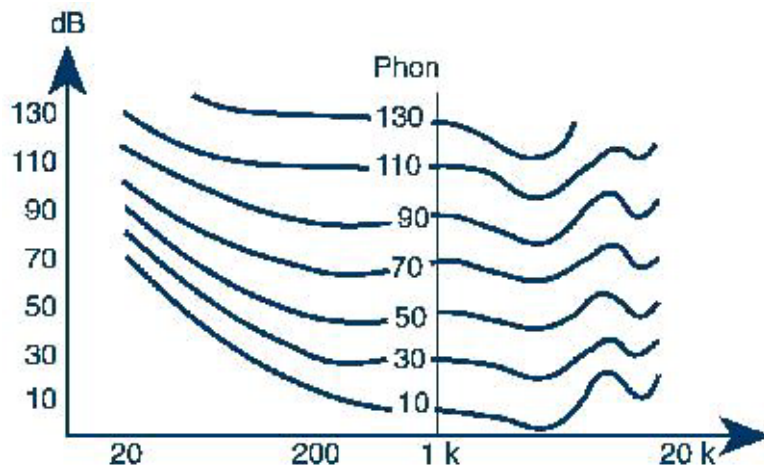
Ear response is even more complicated.

Complex phenomenon to explain.

Illustration : Loudness Curves or Fletcher-Munson Curves:



# What do the curves mean?



- Curves indicate perceived loudness is a function of both the frequency and the level (sinusoidal sound signal)
- Equal loudness curves. Each contour:
  - Equal loudness
  - Express how much a sound level must be changed as the frequency varies,  
**to maintain a certain perceived loudness**

# Physiological Implications

*Why are the curves accentuated where they are?*

- Accentuates of frequency range to coincide with speech.
- Sounds like **p** and **t** have very important parts of their spectral energy within the accentuated range
- Makes them more easy to discriminate between.

The ability to hear sounds of the **accentuated** range (around a few kHz) is thus vital for speech communication.



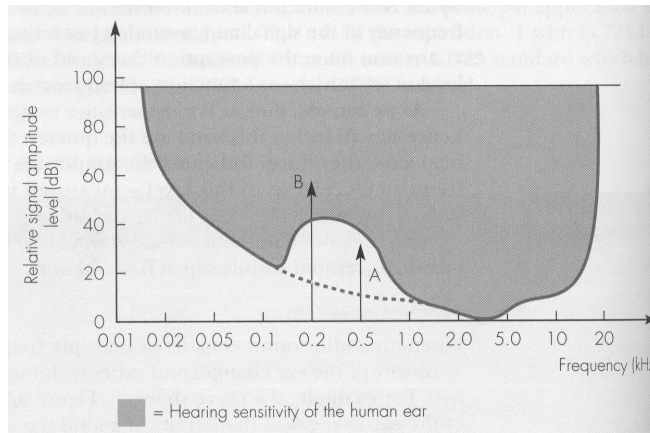
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# Traits of Human Hearing

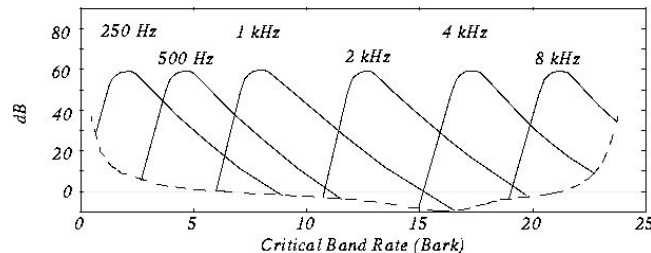
## Frequency Masking

- Multiple frequency audio changes the sensitivity with the relative amplitude of the signals.
- If the frequencies are close and the amplitude of one is less than the other close frequency then the second frequency may not be heard.



# Critical Bands

- Range of closeness for frequency masking depends on the frequencies and relative amplitudes.
- Each **band** where frequencies are masked is called **the Critical Band**
- Critical bandwidth for average human hearing varies with frequency:
  - Constant 100 Hz for frequencies less than 500 Hz
  - Increases (approximately) linearly by 100 Hz for each additional 500 Hz.
- Width of critical band is called a **bark**.





# What is the cause of Frequency Masking?

- **The stereocilia** are excited by air pressure variations, transmitted via outer and middle ear.
- **Different stereocilia** respond to **different ranges** of frequencies — the **critical bands**

**Frequency Masking** occurs because after excitation by one frequency further excitation by a less strong similar frequency of the same group of cells is not possible.

[Click here](#) to hear example of Frequency Masking.



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# Temporal masking

After the ear hears a loud sound:

- It takes a further short while before it can hear a quieter sound.

## Why is this so?

- **Stereocilia** vibrate with corresponding force of input sound stimuli.
- If the stimuli is strong then stereocilia will be in a high state of excitation and **get fatigued**.
- After extended listening to loud music or headphones this sometimes manifests itself with ringing in the ears and even temporary deafness.
- Prolonged exposure to noise permanently damages the **Stereocilia**.

**Temporal Masking** occurs because the hairs take time to settle after excitation to respond again.



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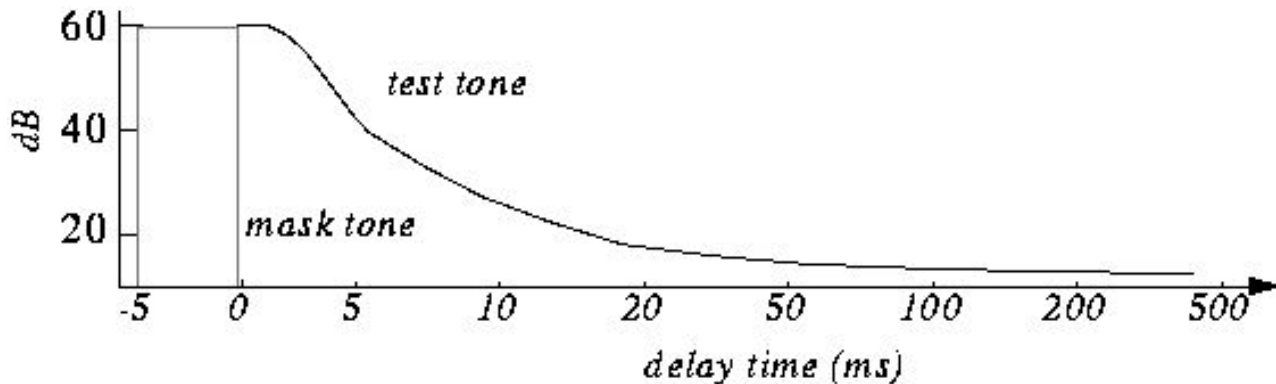
# Example of Temporal Masking

- Play 1 kHz *masking tone* at 60 dB, plus a *test tone* at 1.1 kHz at 40 dB. Test tone can't be heard (it's masked).

Stop masking tone, then stop test tone after a short delay.

Adjust delay time to the shortest time that test tone can be heard (e.g., 5 ms).

Repeat with different level of the test tone and plot:

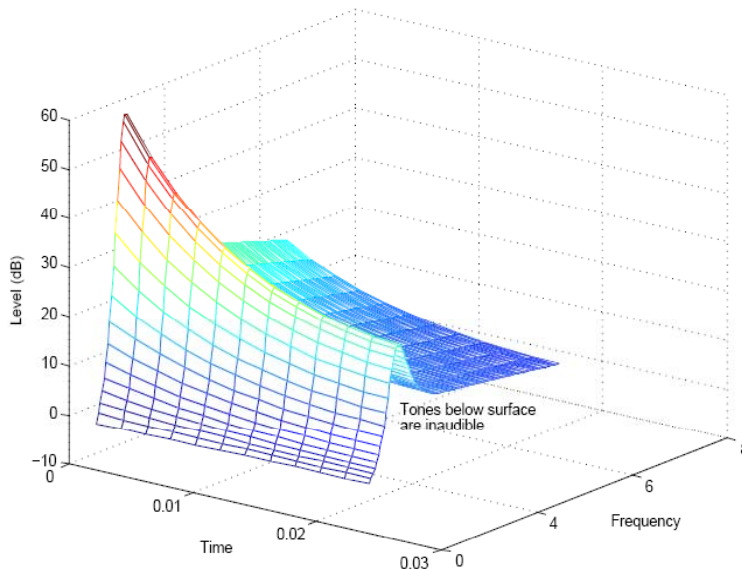


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# Example of Temporal Masking (Cont.)

- Try other frequencies for test tone (masking tone duration constant). Total effect of masking:



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# Compression Idea: How to Exploit?

- If we have a loud tone at, say at 1 kHz, then nearby quieter tones are masked.
- Best compared on **critical band scale** – range of masking is about 1 critical band
- Two factors for masking – **frequency masking** and **temporal masking**
- Question: How to use this for compression?

Two examples:

- **MPEG Audio**
- **Dolby**



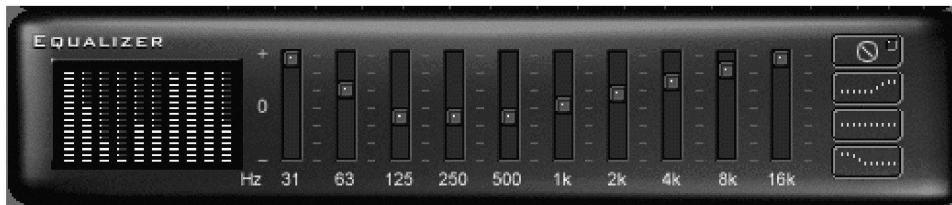
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# How to compute?

We have met basic tools:

- Bank Filtering with **IIR/FIR** Filters
- **Fourier** and **Discrete Cosine Transforms**
  - Work in **frequency space**
- (Critical) **Band Pass Filtering** — Visualise a graphic equaliser



# MPEG Audio Compression

- Exploits the psychoacoustic models above.
- **Frequency masking** is always utilised
- More complex forms of MPEG also employ **temporal masking**.



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# Basic Frequency Filtering Bandpass

MPEG audio compression basically works by:

- Dividing the audio signal up into a set of frequency subbands
- Use Filter Banks to achieve this.
- Subbands approximate **critical bands**.
- Each band quantised according to the **audibility of quantisation noise**.

**Quantisation is the key to MPEG audio compression and is the reason why it is lossy.**



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# How good is MPEG compression?

Although (data) lossy

**MPEG claims** to be **perceptually lossless**:

- Human tests (part of standard development), Expert listeners.
- 6-1 compression ratio, stereo 16 bit samples at 48 Khz compressed to 256 kbits/sec
- **Difficult**, real world examples used.
- **Under Optimal listening conditions** no statistically distinguishable difference between original and MPEG.



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# Basic MPEG: MPEG audio coders

- Set of standards for the use of video **with** sound.
- Compression methods or **coders** associated with audio compression are called **MPEG audio coders**.
- MPEG allows for a variety of different coders to be employed.
- **Difference** in level of sophistication in applying perceptual compression.
- Different **layers** for levels of sophistication.



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# An Advantage of MPEG approach

**Complex psychoacoustic modelling only** in **coding phase**

- Desirable for real time (Hardware or software) decompression
- Essential for broadcast purposes.
- Decompression is independent of the psychoacoustic models used
- Different models can be used
- If there is enough bandwidth no models at all.



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# Basic MPEG: MPEG Standards

Evolving standards for MPEG audio compression:

- MPEG-1 is by the most prevalent
- So called [mp3](#) files we get off Internet are members of [MPEG-1 family](#).
- Standards now extends to MPEG-4 (structured audio) — [Earlier Lecture](#).

[For now we concentrate on MPEG-1](#)



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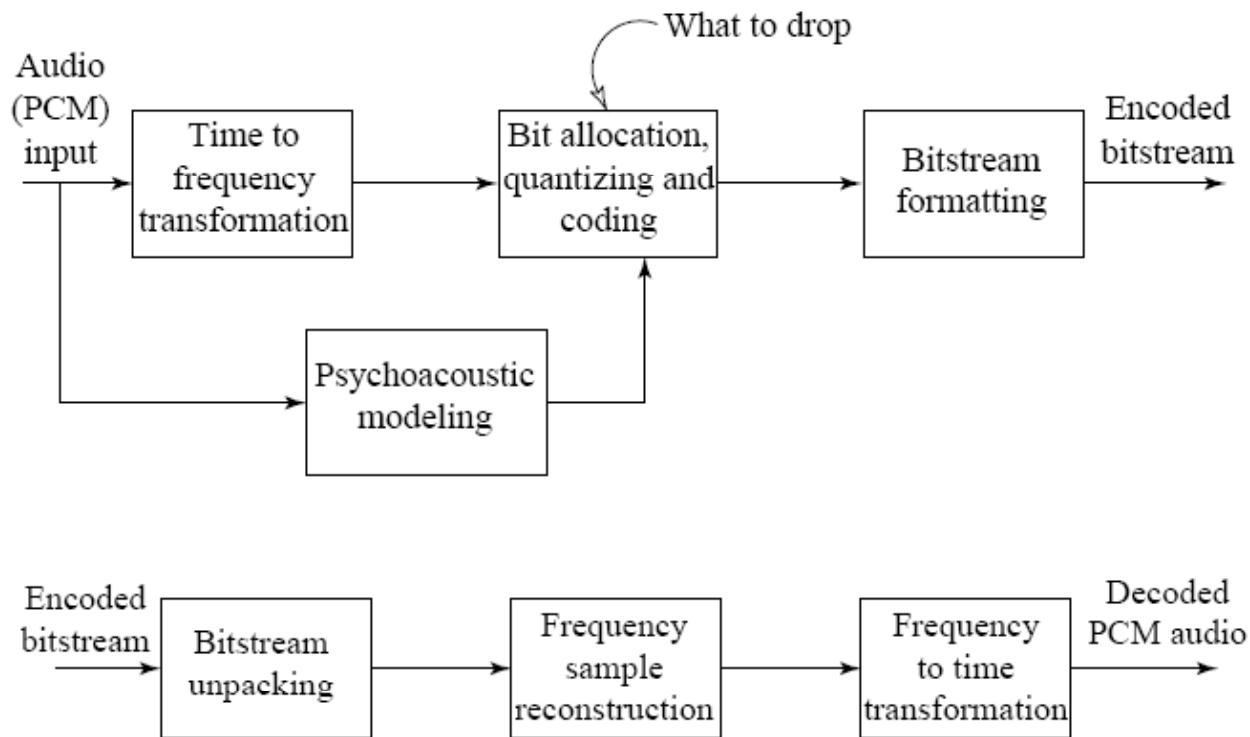
## Basic MPEG: MPEG Facts

- MPEG-1: 1.5 Mbits/sec for audio and video  
About 1.2 Mbits/sec for video, 0.3 Mbits/sec for audio  
(Uncompressed CD audio is  
 $44,100 \text{ samples/sec} * 16 \text{ bits/sample} * 2 \text{ channels} > 1.4 \text{ Mbits/sec}$ )
- Compression factor ranging from 2.7 to 24.
- MPEG audio supports sampling frequencies of 32, 44.1 and 48 KHz.
- Supports one or two audio channels in one of the four modes:
  1. Monophonic – single audio channel
  2. Dual-monophonic – two independent channels  
(functionally identical to stereo)
  3. Stereo – for stereo channels that share bits, but not using joint-stereo coding
  4. Joint-stereo – takes advantage of the correlations between stereo channels



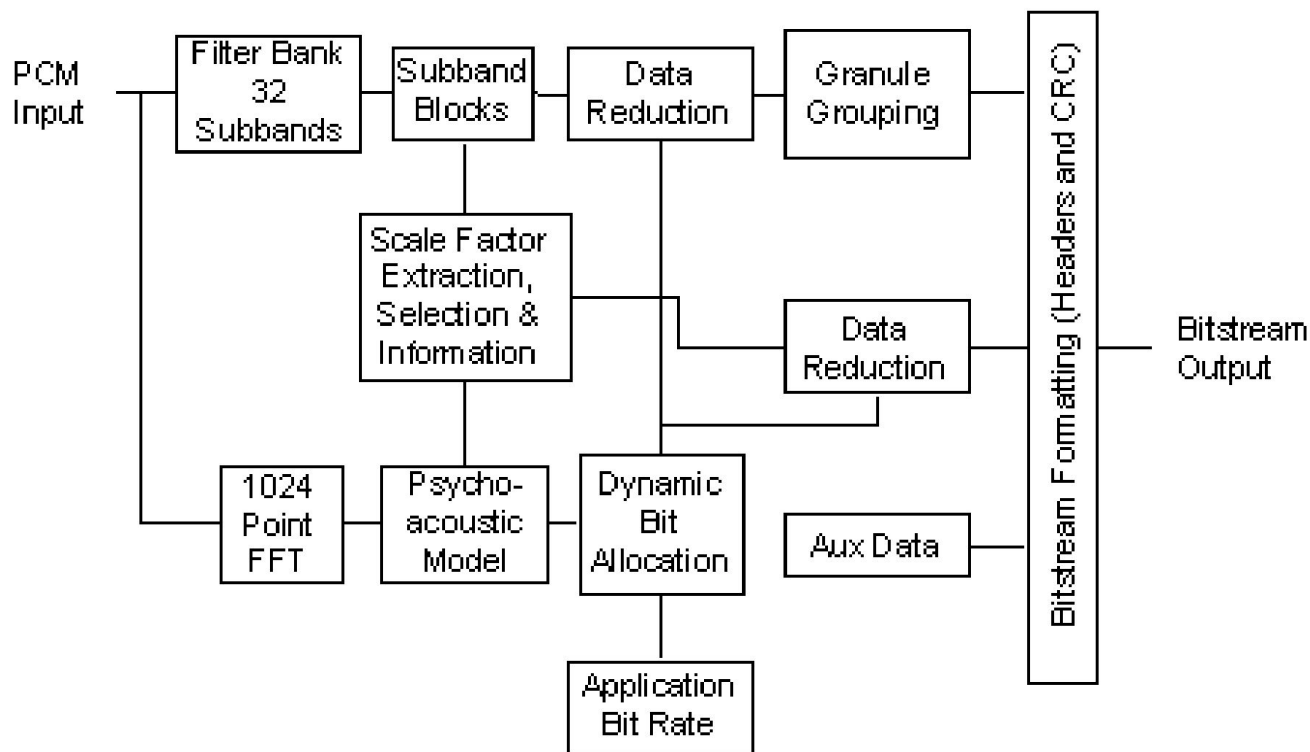
# Basic MPEG-1 Encoding/Decoding algorithm

Basic MPEG-1 encoding/decoding maybe summarised as:



# Basic MPEG-1 Compression algorithm (1)

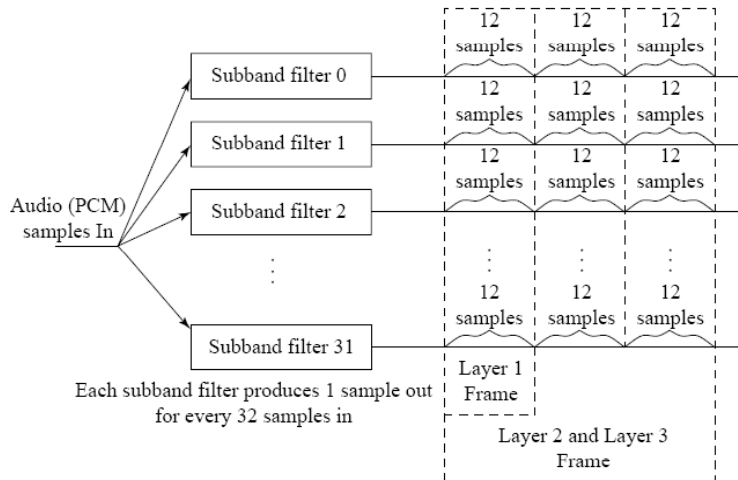
Basic encoding algorithm in more detail summarised below:



# Basic MPEG-1 Compression algorithm (2)

The main stages of the algorithm are:

- The audio signal is first samples and quantised use PCM
  - Application dependent: Sample rate and number of bits
- The PCM samples are then divided up into a number of **frequency subband** and compute **subband scaling factors**:





# Basic MPEG-1 Compression algorithm (3)

## Analysis filters

- Also called **critical-band filters**
- Break signal up into equal width subbands
- Use Filter Banks (modified with discrete cosine transform (DCT) Level 3)
- Filters divide audio signal into frequency subbands that approximate the 32 critical bands
- Each band is known as a *sub-band sample*.
- **Example:** 16 kHz signal frequency, Sampling rate 32 kbits/sec gives each subband a bandwidth of 500 Hz.
- Time duration of each sampled segment of input signal is time to accumulate 12 successive sets of 32 PCM (subband) samples, **i.e.**  $32 \times 12 = 384$  samples.



# Basic MPEG-1 Compression algorithm (4)

## Analysis filters (cont)

- In addition to filtering the input, analysis banks determine
  - Maximum amplitude of 12 subband samples in each subband.
  - Each known as the **scaling factor** of the subband.
  - Passed to *psychoacoustic model* and **quantiser blocks**



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# Basic MPEG-1 Compression algorithm (5)

## *Psychoacoustic modeller:*

- Frequency Masking and may employ temporal masking.
- Performed concurrently with filtering and analysis operations.
- Uses Fourier Transform (FFT) to perform analysis.
- Determine amount of masking for each band caused by nearby bands.
- Input: set hearing thresholds and subband masking properties (model dependent) and scaling factors (above).



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# Basic MPEG-1 Compression algorithm (6)

## *Psychoacoustic modeller (cont):*

- Output: a set of **signal-to-mask** ratios:
  - Indicate those frequencies components whose amplitude is below the audio threshold.
  - If the power in a band is below the masking threshold, don't encode it.
  - Otherwise, determine number of bits (from scaling factors) needed to represent the coefficient such that noise introduced by quantisation is below the masking effect (Recall that 1 bit of quantisation introduces about 6 dB of noise).



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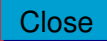
# Basic MPEG-1 Compression algorithm (7)

## Example of Quantisation:

- Assume that after analysis, the first levels of 16 of the 32 bands are:

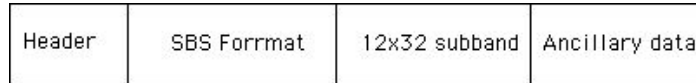
Band	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Level (db)	0	8	12	10	6	2	10	60	35	20	15	2	3	5	3	1

- If the level of the 8th band is 60 dB,  
then assume (according to model adopted) it gives a masking of  
12 dB in the 7th band, 15 dB in the 9th.  
Level in 7th band is 10 dB (  $< 12$  dB ), so ignore it.  
Level in 9th band is 35 dB (  $> 15$  dB ), so send it.  
→ Can encode with up to 2 bits (= 12 dB) of quantisation error.
- More on Bit Allocation soon.**



# MPEG-1 Output bitstream

The basic output stream for a basic MPEG encoder is as follows:



- **Header**: contains information such as the sample frequency and quantisation,.
- **Subband sample (SBS) format**: Quantised scaling factors and 12 frequency components in each subband.
  - Peak amplitude level in each subband quantised using 6 bits (64 levels)
  - 12 frequency values quantised to 4 bits
- **Ancillary data**: Optional. Used, for example, to carry additional coded samples associated with special broadcast format (**e.g surround sound**)

# Decoding the bitstream

- Dequantise the subband samples after demultiplexing the coded bitstream into subbands.
- **Synthesis bank** decodes the dequantised subband samples to produce PCM stream.
  - This essentially involves applying the inverse fourier transform (**IFFT**) on each substream and multiplexing the channels to give the PCM bit stream.



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# MPEG Layers

MPEG defines 3 levels of processing layers for audio:

- Level 1 is the basic mode,
- Levels 2 and 3 more advance (use temporal masking).
- Level 3 is the most common form for audio files on the Web
  - Our beloved MP3 files that record companies claim are bankrupting their industry.
  - Strictly speaking these files should be called **MPEG-1 level 3** files.

Each level:

- Increasing levels of sophistication
- Greater compression ratios.
- Greater computation expense



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# Level 1

- Best suited for bit rate bigger than 128 kbits/sec per channel.
- Example: Phillips Digital Compact Cassette uses Layer 1 192 kbits/sec compression
- Divides data into frames,
  - Each of them contains 384 samples,
  - 12 samples from each of the 32 filtered subbands as shown above.
- Psychoacoustic model only uses frequency masking.
- Optional Cyclic Redundancy Code (CRC) error checking.



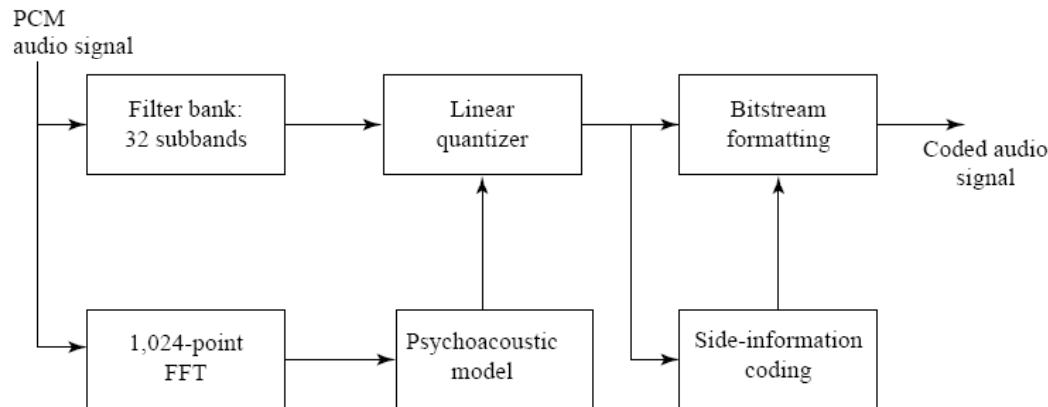
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# Level 1 (and Level 2) Audio Layers

**Note:** Mask Calculations done in parallel with subband filtering

- Accurate frequency decomposition via Fourier Transform.



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# Layer 2

- Targeted at bit rates of around 128 kbits/sec per channel.
- Examples: Coding of Digital Audio Broadcasting (DAB) on CD-ROM, CD-I and Video CD.
- Enhancement of level 1.
- Codes audio data in larger groups:
  - Use three frames in filter:  
**before, current, next**, a total of 1152 samples.
  - This models a little bit of the temporal masking.
- Imposes some restrictions on bit allocation in middle and high subbands.
- More compact coding of scale factors and quantised samples.
- Better audio quality due to saving bits here so more bits can be used in quantised subband values



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# Layer 3

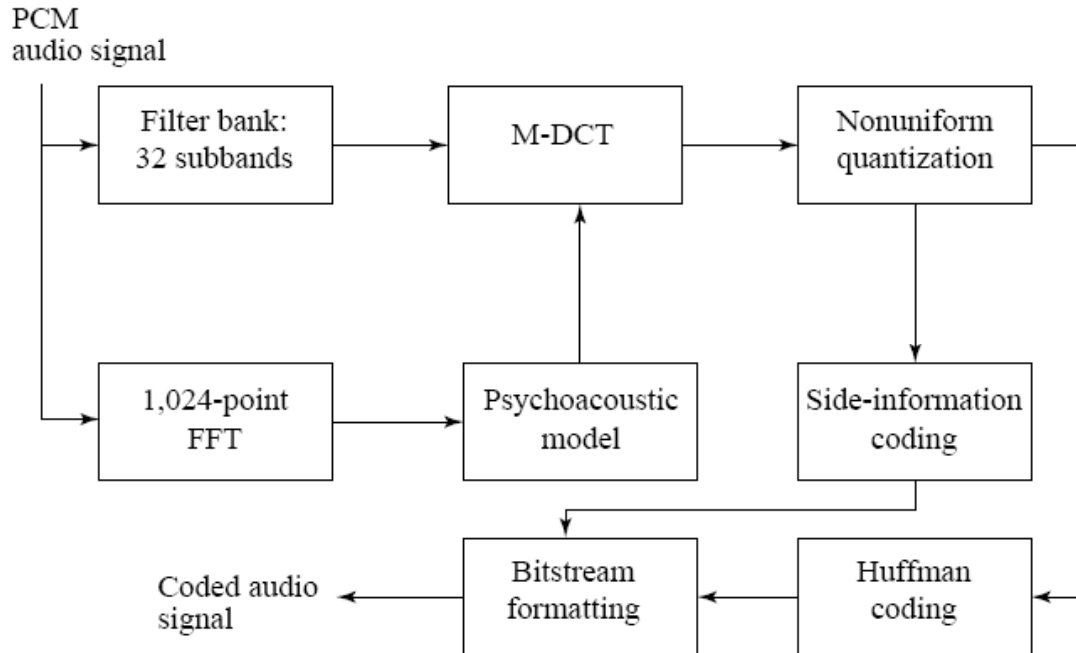
- Targeted at bit rates of 64 kbits/sec per channel.
- Example: audio transmission of ISDN or suitable bandwidth network.
- Much more complex approach.
- Psychoacoustic model includes temporal masking effects,
- Takes into account stereo redundancy.
- Better critical band filter is used (non-equal frequencies)
- Uses a modified DCT (MDCT) for lossless subband transformation.
- Two different block lengths: 18 (long) or 6 (short)
- 50% overlap between successive transform windows gives window sizes of 36 or 12 — **accounts for temporal masking**
- Greater frequency resolution accounts for poorer time resolution
- Uses Huffman coding on quantised samples for better compression.



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# Level 3 Audio Layers



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# Bit Allocation

- Process determines the number of code bits for each subband
- Based on information from the psychoacoustic model.



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# Bit Allocation For Layer 1 and 2

- Compute the mask-to-noise ratio (MNR) for all subbands:

$$MNR_{dB} = SNR_{dB} - SMR_{dB}$$

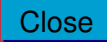
where

$MNR_{dB}$  is the mask-to-noise ratio,

$SNR_{dB}$  is the signal-to-noise ratio (SNR), and

$SMR_{dB}$  is the signal-to-mask ratio from the psychoacoustic model.

- Standard MPEG lookup tables estimate SNR for given quantiser levels.
- Designers are free to try other methods SNR estimation.



# Bit Allocation For Layer 1 and 2 (cont.)

Once MNR computed for all the subbands:

- Search for the subband with the lowest MNR
- Increment code bits to that subband.
- When a subband gets allocated more code bits, the bit allocation unit:
  - Looks up the new estimate for SNR
  - Recomputes that subband's MNR.
- The process repeats until no more code bits can be allocated.

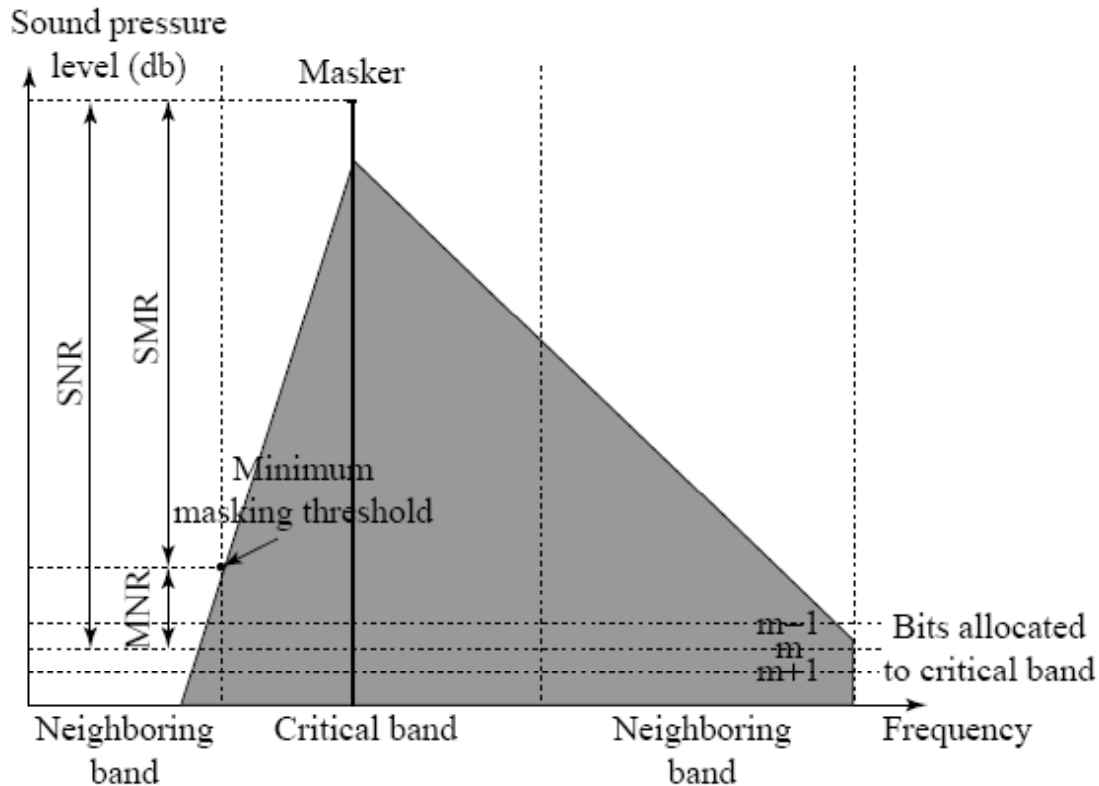


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# Bit Allocation For Layer 1 and 2 (cont.)



# Bit Allocation For Layer 3

- Uses **noise allocation**, which employs Huffman coding.
- Iteratively varies the quantisers in an orderly way
  - Quantises the spectral values,
  - Counts the number of Huffman code bits required to code the audio data
  - Calculates the resulting noise in Huffman coding.

If there exist scale factor bands with more than the allowed distortion:

  - Encoder amplifies values in bands
  - **To effectively decreases** the quantiser step size for those bands.



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# Bit Allocation For Layer 3 (Cont.)

After this the process repeats. The process stops if any of these three conditions is true:

- None of the scale factor bands have more than the allowed distortion.
- The next iteration would cause the amplification for any of the bands to exceed the maximum allowed value.
- The next iteration would require all the scale factor bands to be amplified.

Real-time encoders include a time-limit exit condition for this process.

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# Stereo Redundancy Coding

## Exploit redundancy in two couple stereo channels?

- Another perceptual property of the human auditory system
- Simply stated at low frequencies, the human auditory system can't detect where the sound is coming from.
  - So save bits and encode it mono.
- **Used in MPEG-1 Layer 3.**

Two types of stereo redundancy coding:

- Intensity stereo coding — **all layers**
- Middle/Side (MS) stereo coding — **Layer 3 only** stereo coding.



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# Intensity stereo coding

Encoding:

- Code some upper-frequency subband outputs:
  - A single summed signal instead of sending independent left and right channels codes
  - Codes for each of the 32 subband outputs.

Decoding:

- Reconstruct left and right channels
  - Based only on a single summed signal
  - Independent left and right channel scale factors.

With intensity stereo coding,

- The spectral shape of the left and right channels is the same within each intensity-coded subband
- **But** the magnitude is different.



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# Middle/Side (MS) stereo coding

- Encodes the left and right channel signals in certain frequency ranges:
  - **Middle** — sum of left and right channels
  - **Side** — difference of left and right channels.
- Encoder uses specially tuned threshold values to compress the side channel signal further.

## MATLAB MPEG Audio Coding Code

[MPEGAudio](#) (DIRECTORY)

[MPEGAudio.zip](#) (All Files Zipped)



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# Dolby Audio Compression

Application areas:

- FM radio Satellite transmission and broadcast TV audio (DOLBY AC-1)
- Common compression format in PC sound cards (DOLBY AC-2)
- High Definition TV standard **advanced television** (ATV) (DOLBY AC-3). *MPEG a competitor in this area.*



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# Differences with MPEG

- MPEG perceptual coders control quantisation accuracy of each subband by computing bit numbers for each sample.
- MPEG needs to store each quantise value with each sample.
- MPEG Decoder uses this information to dequantise:  
**forward adaptive bit allocation**
- **Advantage of MPEG?**: no need for psychoacoustic modelling in the decoder due to store of every quantise value.
- **DOLBY**: Use **fixed bit rate allocation** for each subband.
  - No need to send with each frame — as in **MPEG**.
  - **DOLBY** encoders and decoder need this information.



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# Fixed Bit Rate Allocation

- Bit allocations are determined by known sensitivity characteristics of the ear.



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# Different Dolby standards

## DOLBY AC-1 :

Low complexity psychoacoustic model

- 40 subbands at sampling rate of 32 kbits/sec or
- (Proportionally more) Subbands at 44.1 or 48 kbits/sec
- Typical compressed bit rate of 512 kbits per second for stereo.
- Example: FM radio Satellite transmission and broadcast TV audio



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## DOLBY AC-2 :

Variation to allow subband bit allocations to vary

- **NOW** Decoder needs copy of psychoacoustic model.
- Minimised encoder bit stream overheads at expense of transmitting encoded frequency coefficients of sampled waveform segment — known as the **encoded spectral envelope**.
- Mode of operation known as **backward adaptive bit allocation mode**
- High (hi-fi) quality audio at 256 kbits/sec.
- Not suited for broadcast applications:
  - encoder cannot change model without changing (remote/distributed) decoders
- Example: Common compression format in PC sound cards.



## DOLBY AC-3 :

Development of AC-2 to overcome broadcast challenge

- Use **hybrid backward/forward adaptive bit allocation mode**
- Any model modification information is encoded in a frame.
- Sample rates of 32, 44.1, 48 kbits/sec supported depending on bandwidth of source signal.
- Each encoded block contains 512 subband samples, with 50% (256) overlap between successive samples.
- For a 32 kbits/sec sample rate each block of samples is of 8 ms duration, the duration of each encoder is 16 ms.
- Audio bandwidth (at 32 kbits/sec) is 15 KHz so each subband has 62.5 Hz bandwidth.
- Typical stereo bit rate is 192 kbits/sec.
- Example: High Definition TV standard **advanced television** (ATV). MPEG competitor in this area.



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That's All Folks

At Last

The End of CM0340!!

Happy Christmas



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