BH001-2 : System Solution

Multimedia Systems

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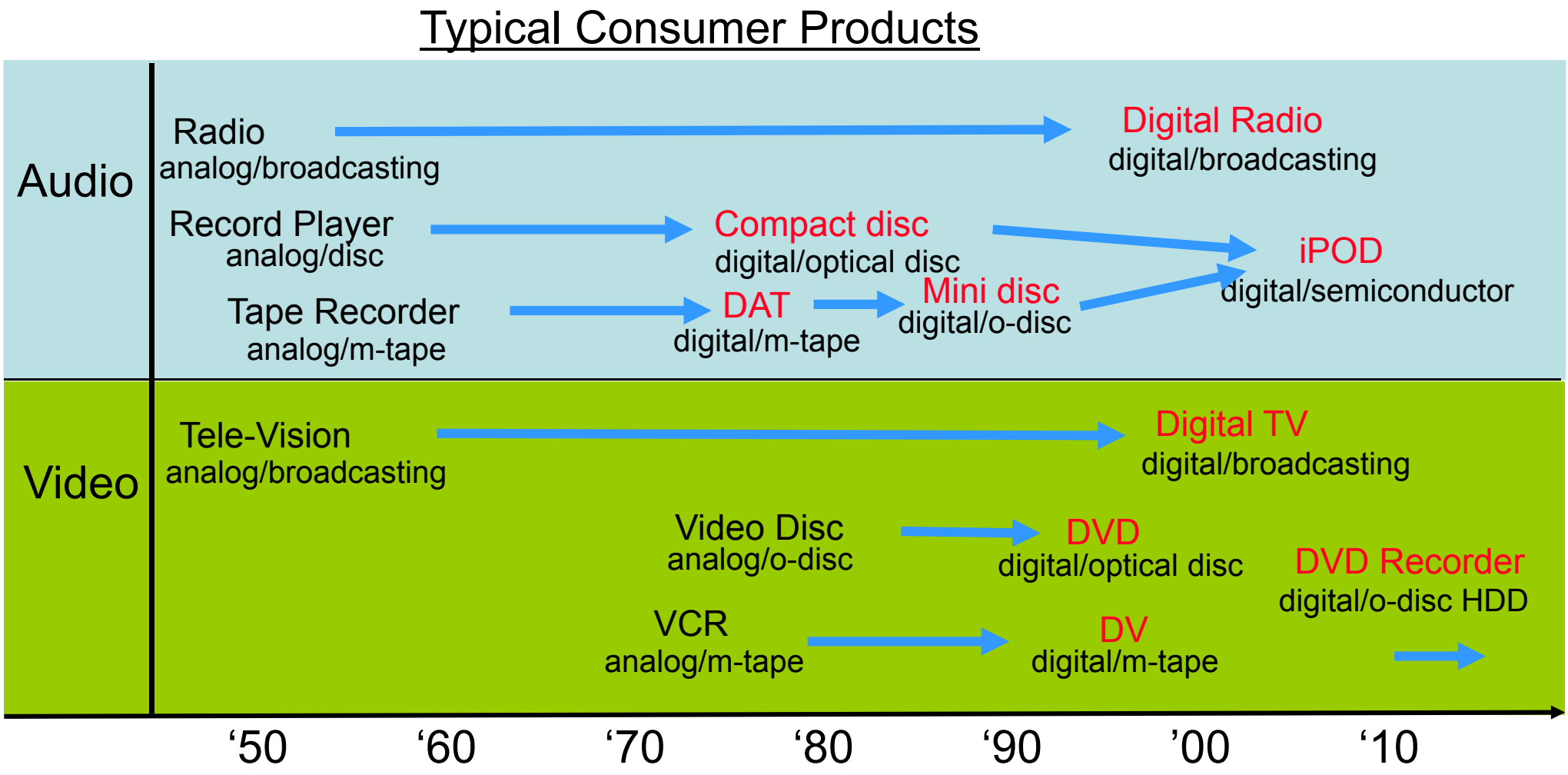
AGENDA

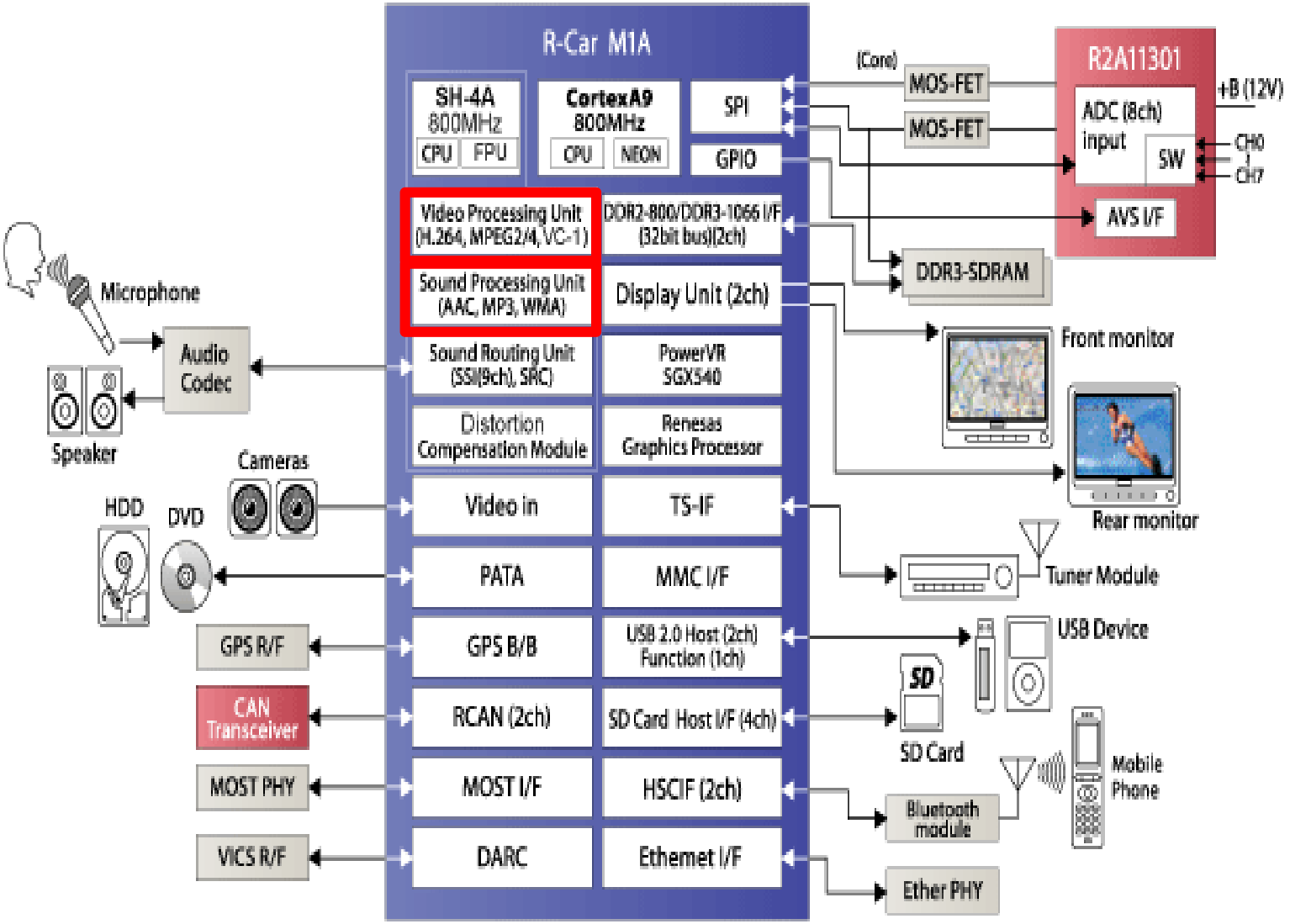
- Audio signal processing & Audio codec
- Video signal processing & Video codec
- Example of SoC architecture

Trends of AV signal processing technology

AV equipments have migrated from analog technology to digital technology



Multimedia Processing in Renesas's SoC

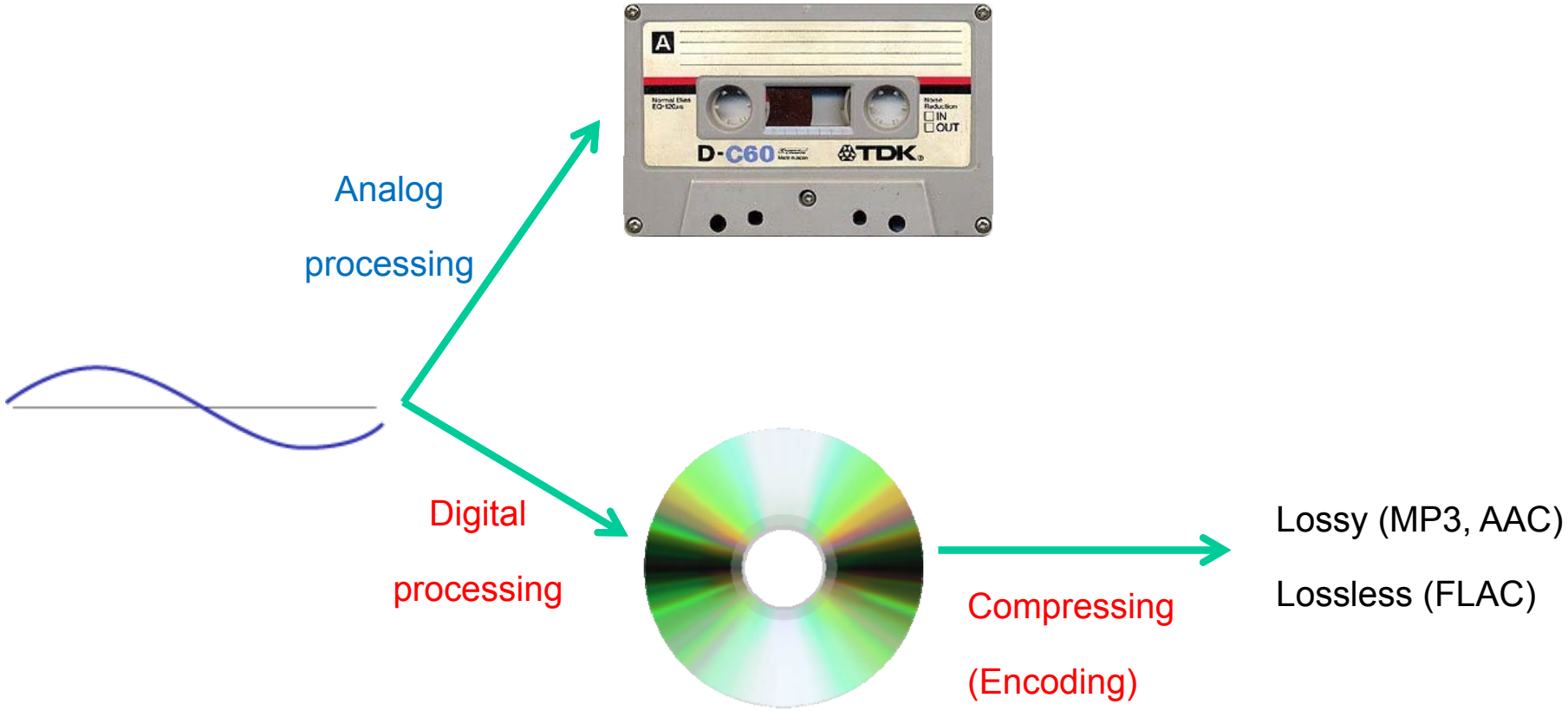


Audio signal processing & Audio codec

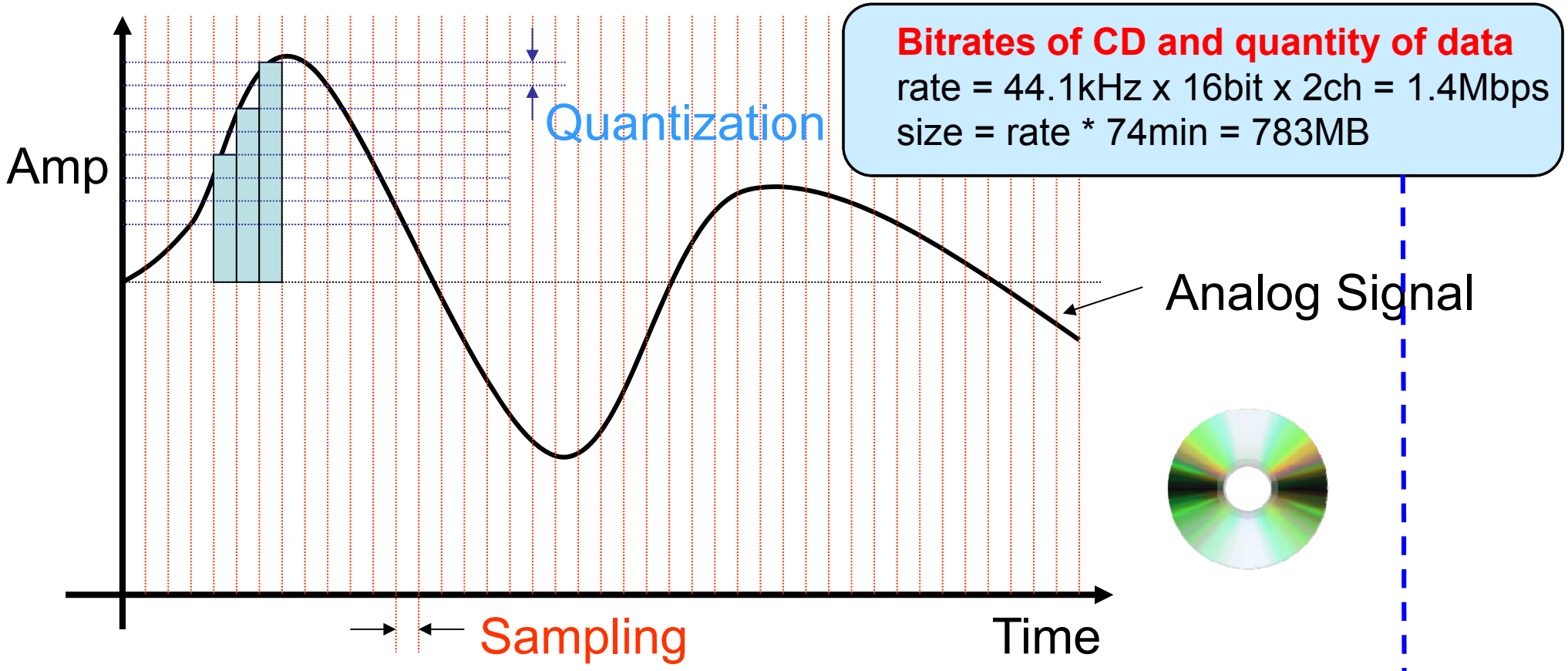
Content

- Introduction
- Basic technique of Audio Coding (Loosy & Lossless)
- Basic technique of LOOSY Audio Coding (Psych-acoustic Model)

Trends of Audio processing



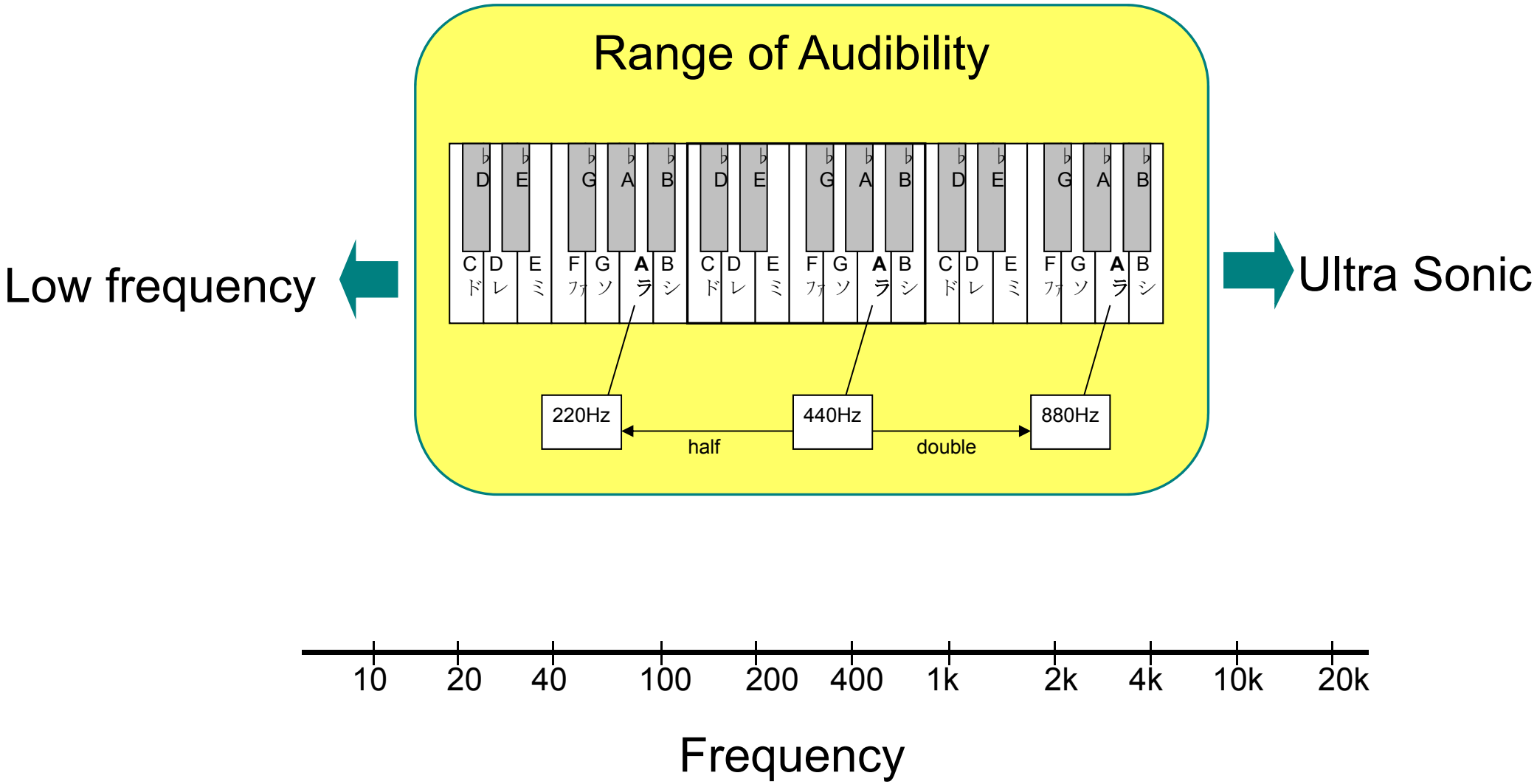
Digital signal of audio and voice



Bitrates of CD and quantity of data
rate = 44.1kHz x 16bit x 2ch = 1.4Mbps
size = rate * 74min = 783MB

A large quantity of information is needed handing digitalized Audio and especially Video Signal → Demand for compression method.

Frequency of audio and voice signal



Variety of sound signal processing

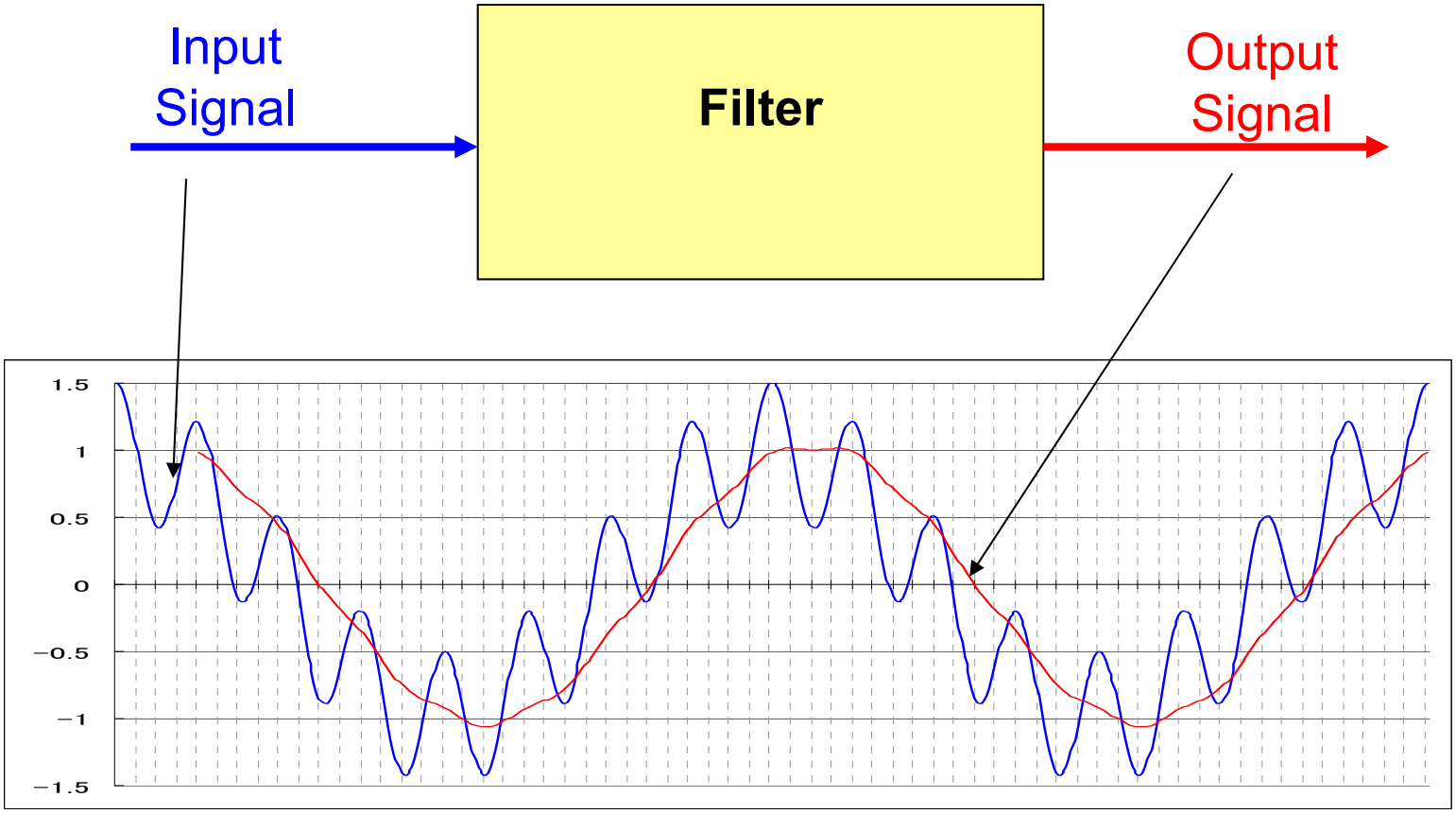
Forming waves → Filter Processing

- Equalizer
- Noise Canceller
- Rate Conversion

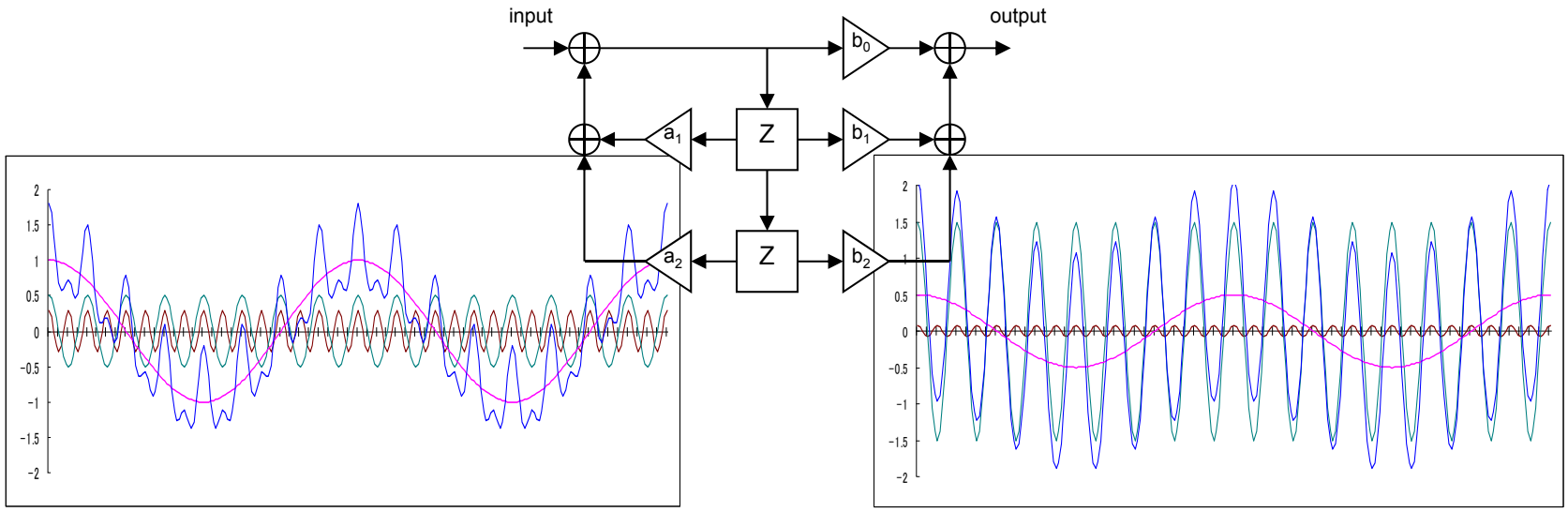
Data Compression → Standard Coding

- Voice Codec standard
- Audio Coding standard

Filter processing

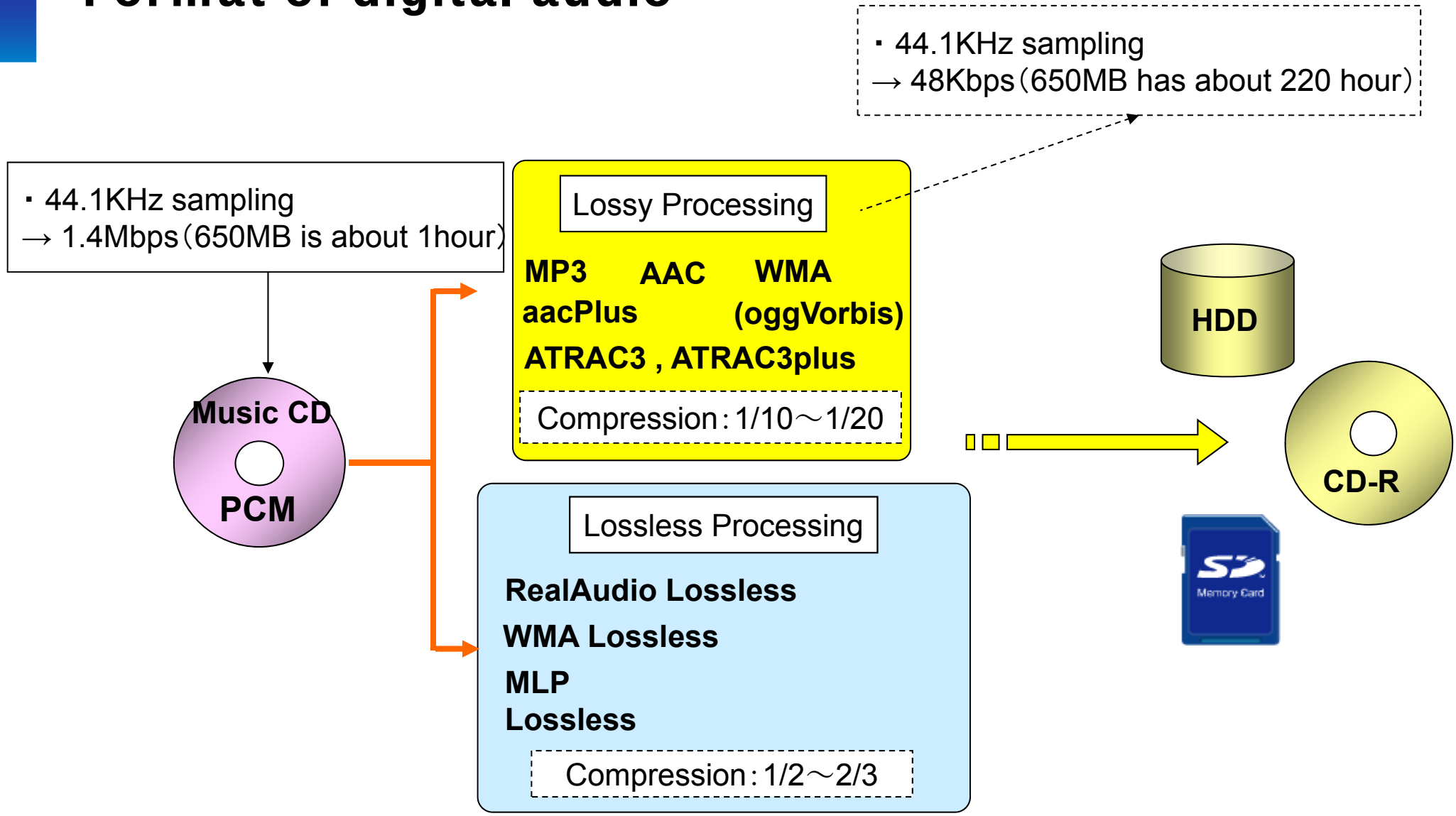


Example Equalizer



Adjust Treble or Bass

Format of digital audio



Basic principle of Audio Coding

(1) Utilize Time – Frequency transformation

— Fourier transformation
(MDCT – Modifed Discrete CosineTransform)

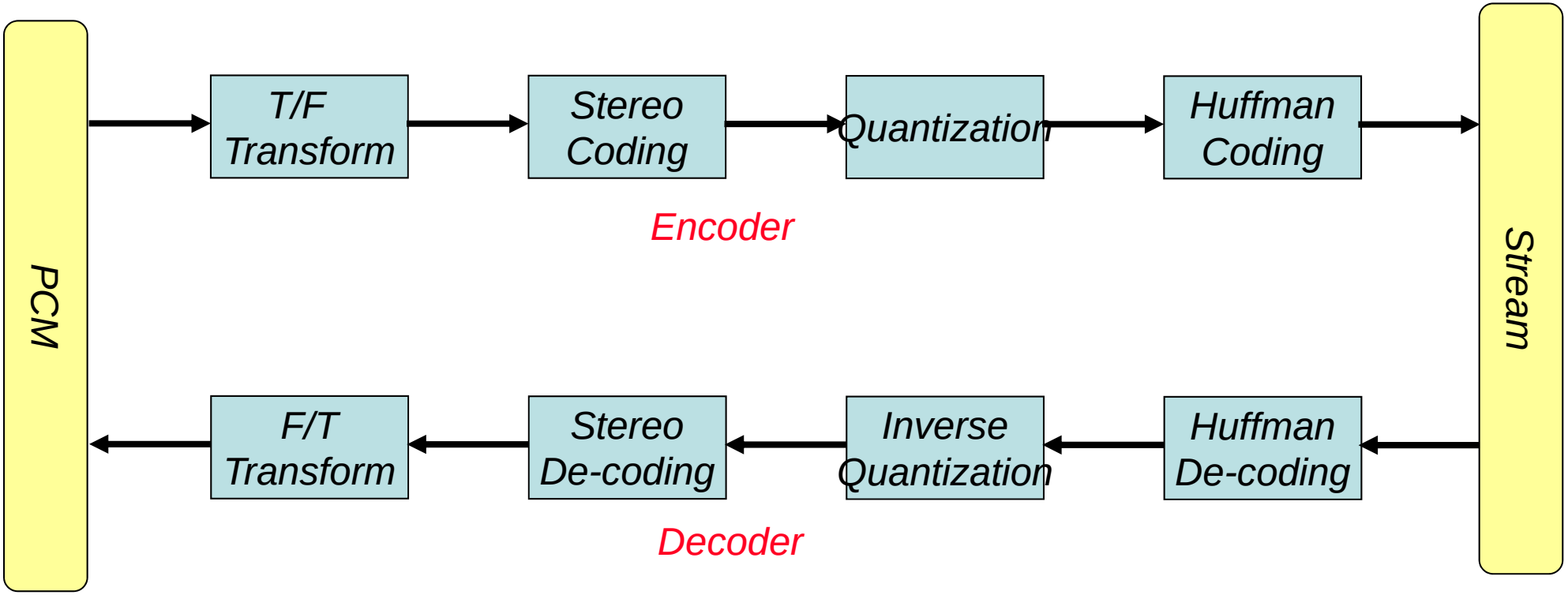
(2) Utilize the property of correlation between Audio data

— Stereo Coding

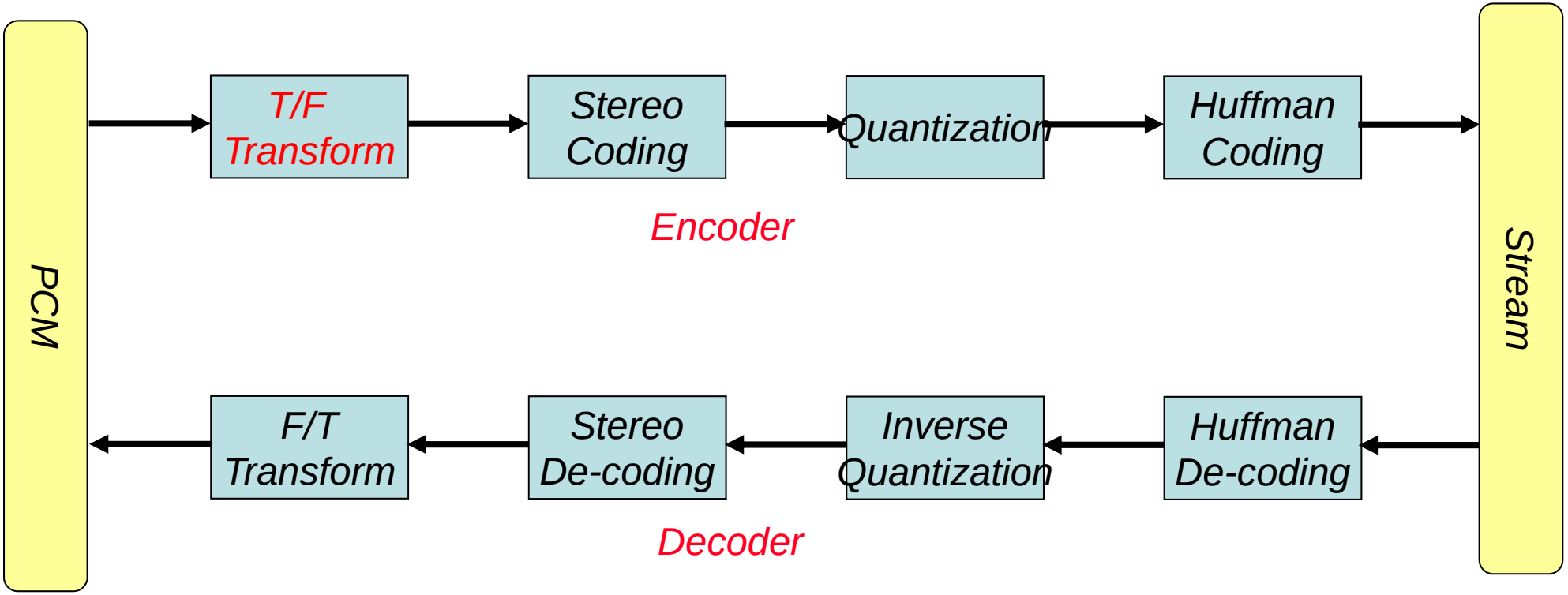
(3) Utilize the probability of data appearance

— Entropy Coding
（Huffman Coding）

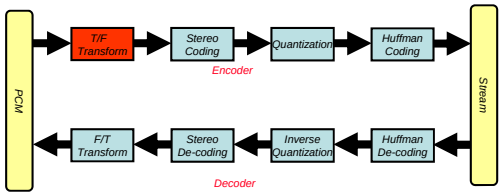
Basic block diagram of audio codec



Basic block diagram of audio codec



Time-Frequency transformation



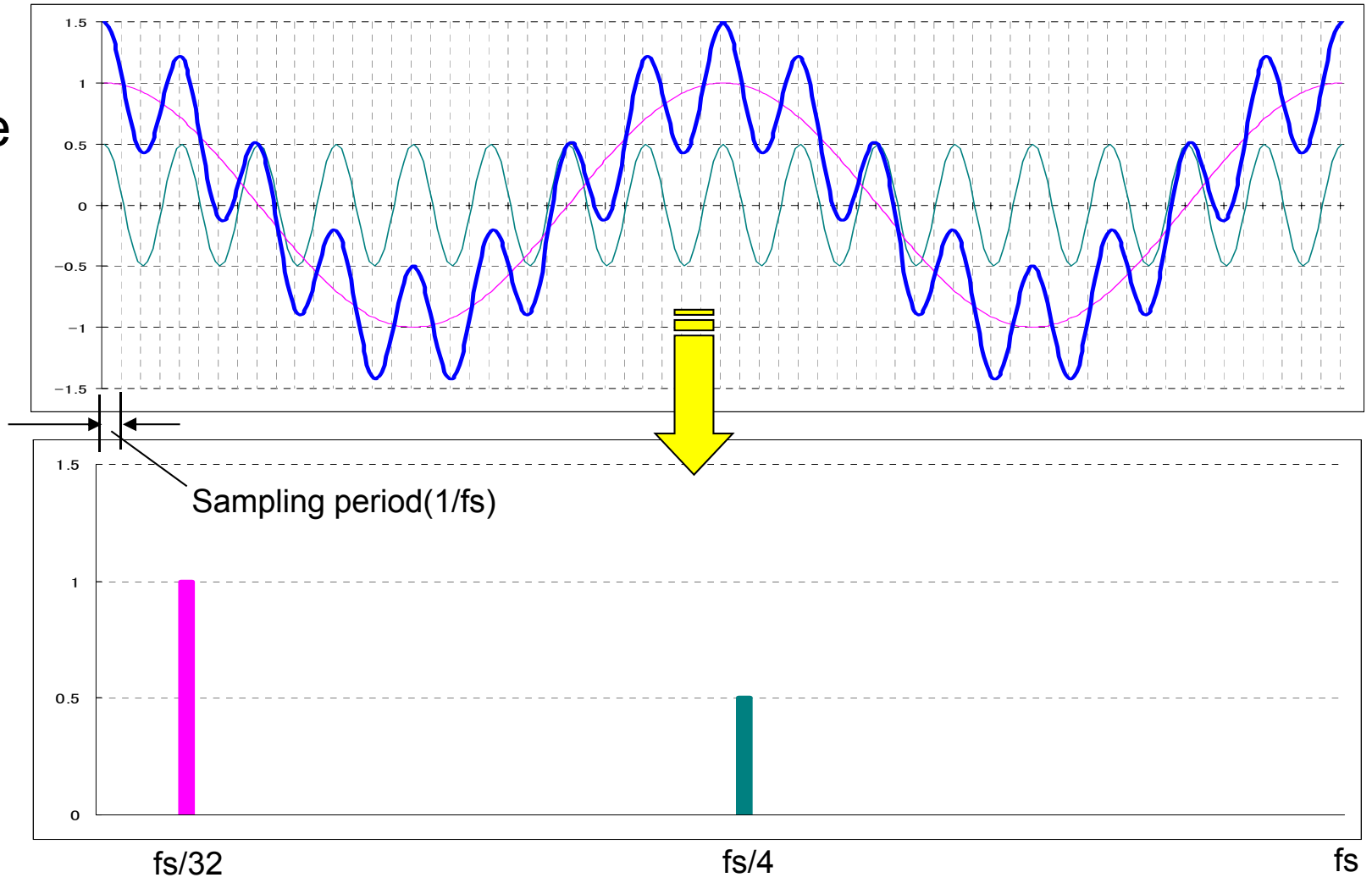
Audio signal can be transferred to frequency domain by using QMF Filter Bank or MDCT

Time Scale

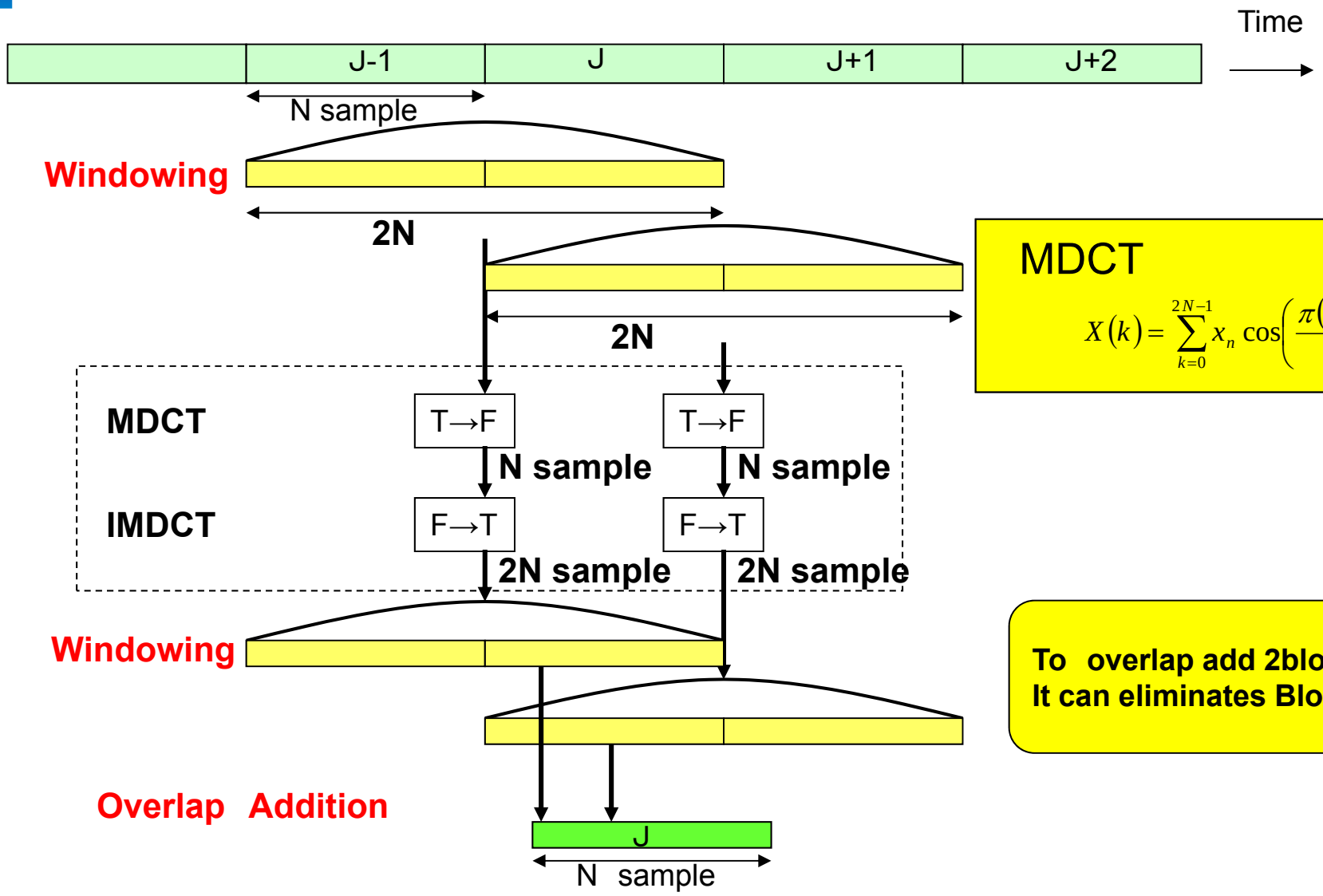
PCM data

Coefficient data

Frequency Scale



MDCT transformation

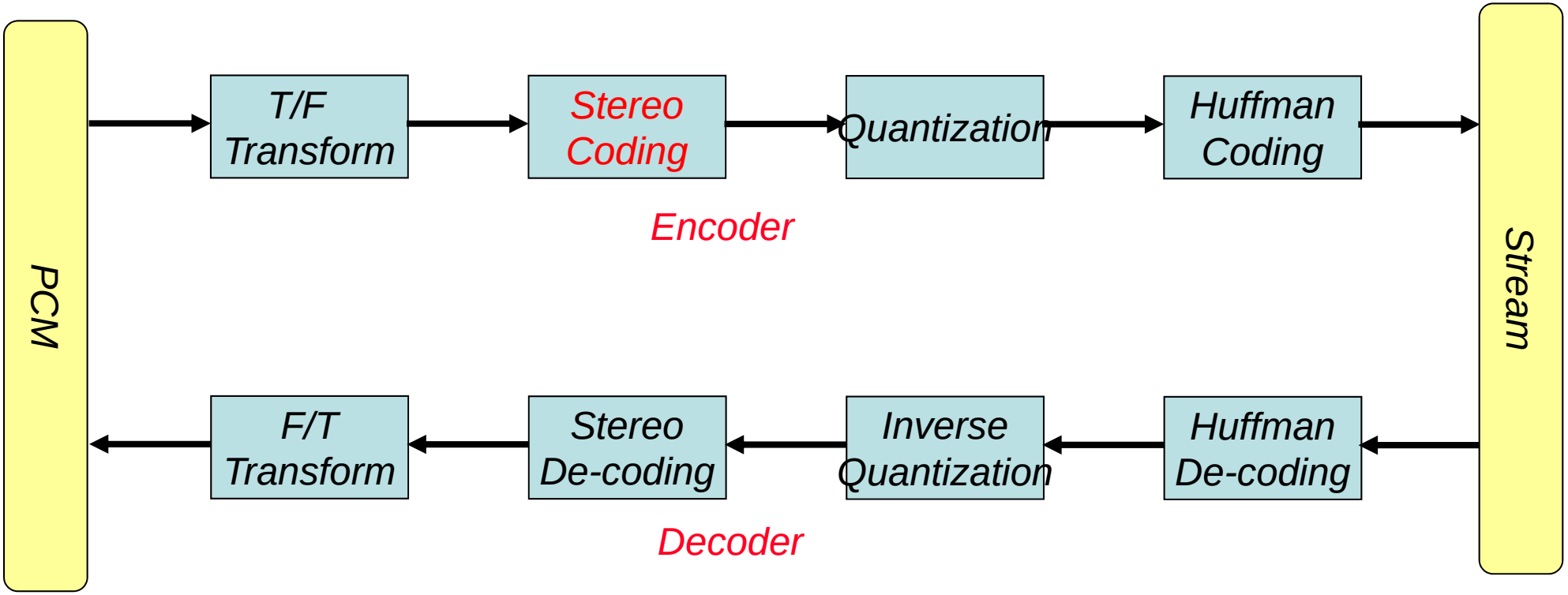


MDCT

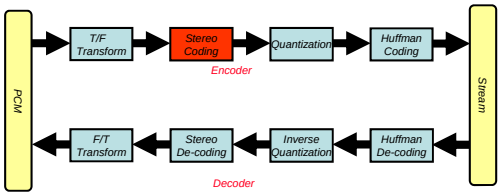
$$X(k) = \sum_{k=0}^{2N-1} x_n \cos\left(\frac{\pi(2k+1)(2n+N+1)}{4M}\right)$$

To overlap add 2bloks with windowing, It can eliminates Block distortion.

Basic block diagram of audio codec



MS-Stereo (Stereo coding)

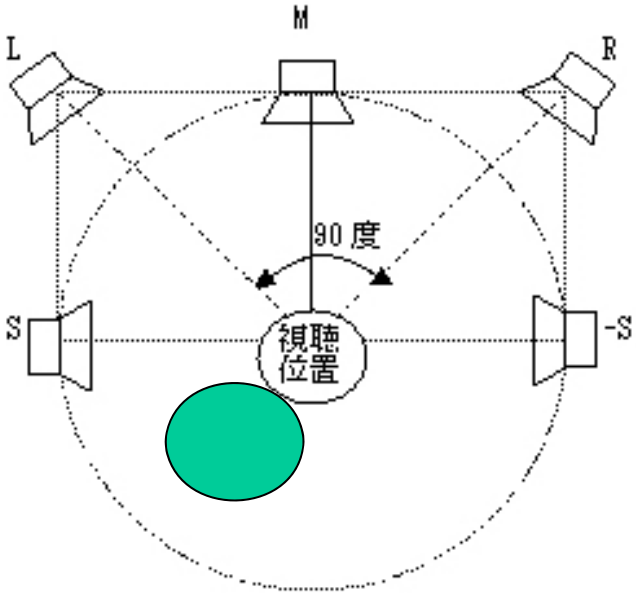


Use correlation between Left and Right channel in stereo audio data

MS(Middle/Side) Stereo

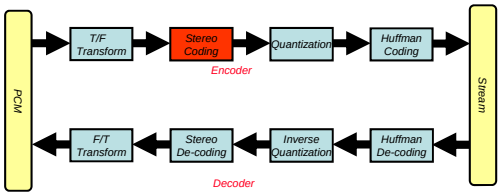
Sum of 2ch component M=(L＋R)

Difference of 2ch component S=(L—R)



⟶ Suppose the difference between L-ch and R-ch is very small, S close to 0. It contribute decrease the quantity of data.

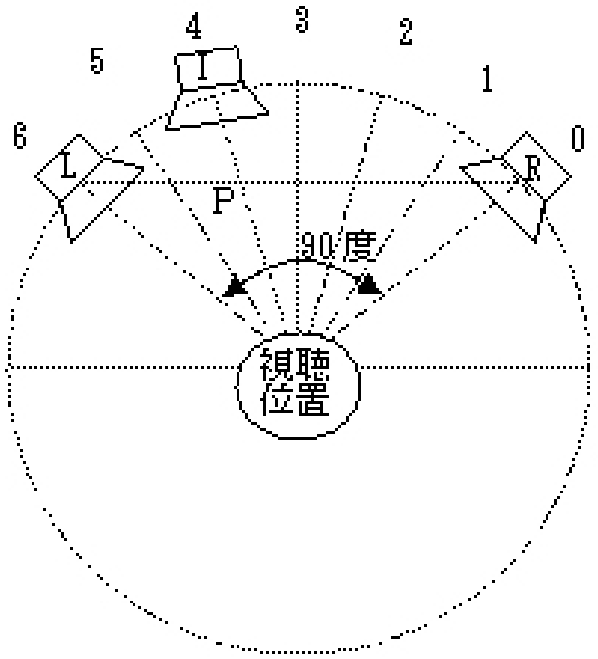
Intensity stereo (Stereo coding)



Use correlation between Left and Right channel in stereo audio data

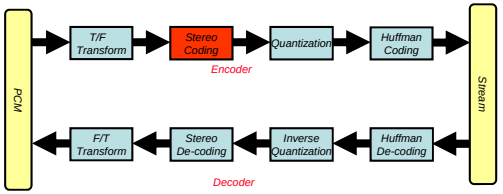
Intensity Stereo

Common Signal	S
Ratio of L/R	is_ratio



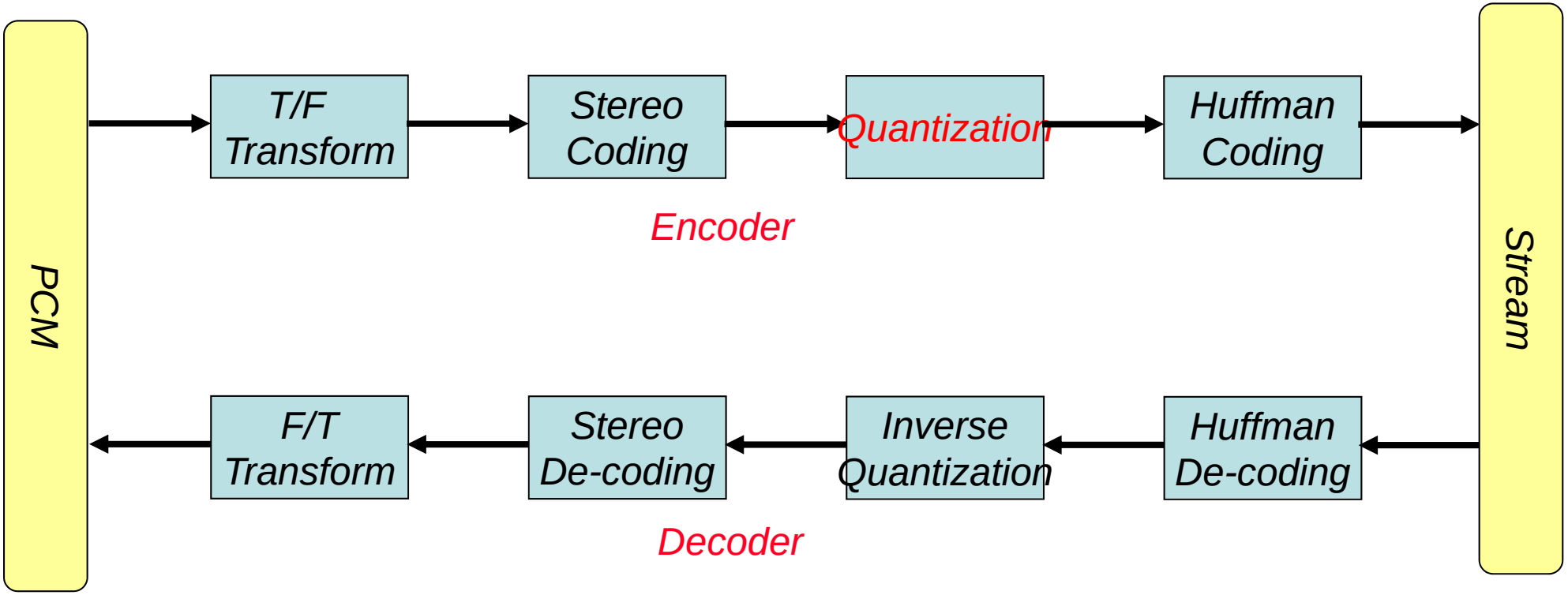
→ For high frequency signal, Amplitude of each channel are important compare with the Signal phase difference.

Joint stereo
(Combination of stereo coding)

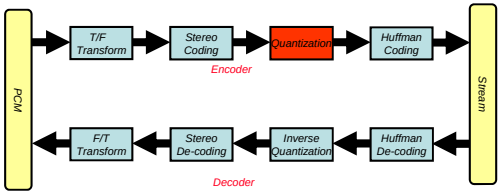


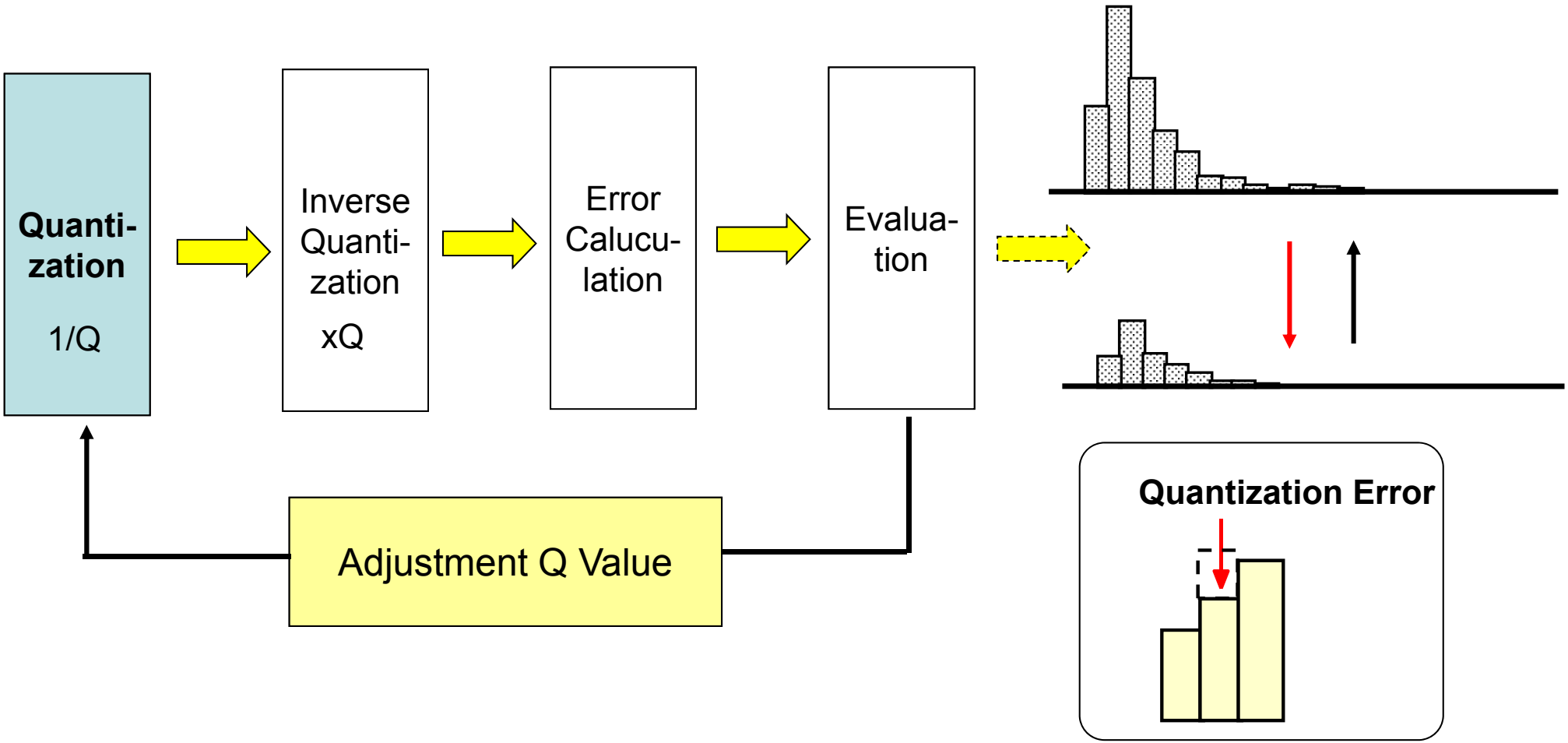
	MS stereo	Intensity stereo
Lch	Component M	Component S
Rch	Component S	Component position

Basic block diagram of audio codec

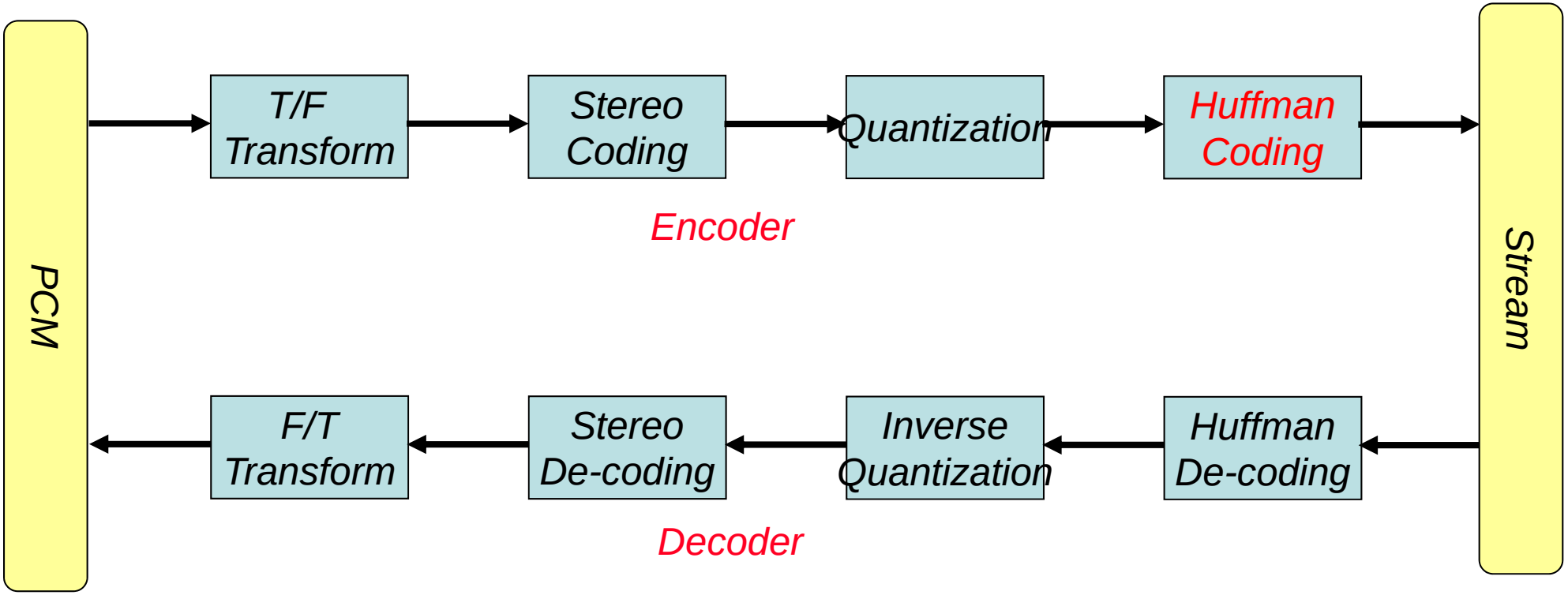


Quantization

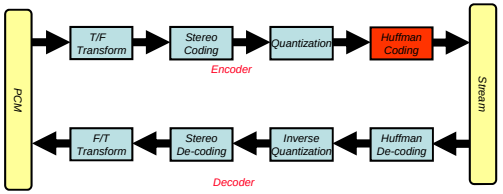




Basic block diagram of audio codec



Huffman coding



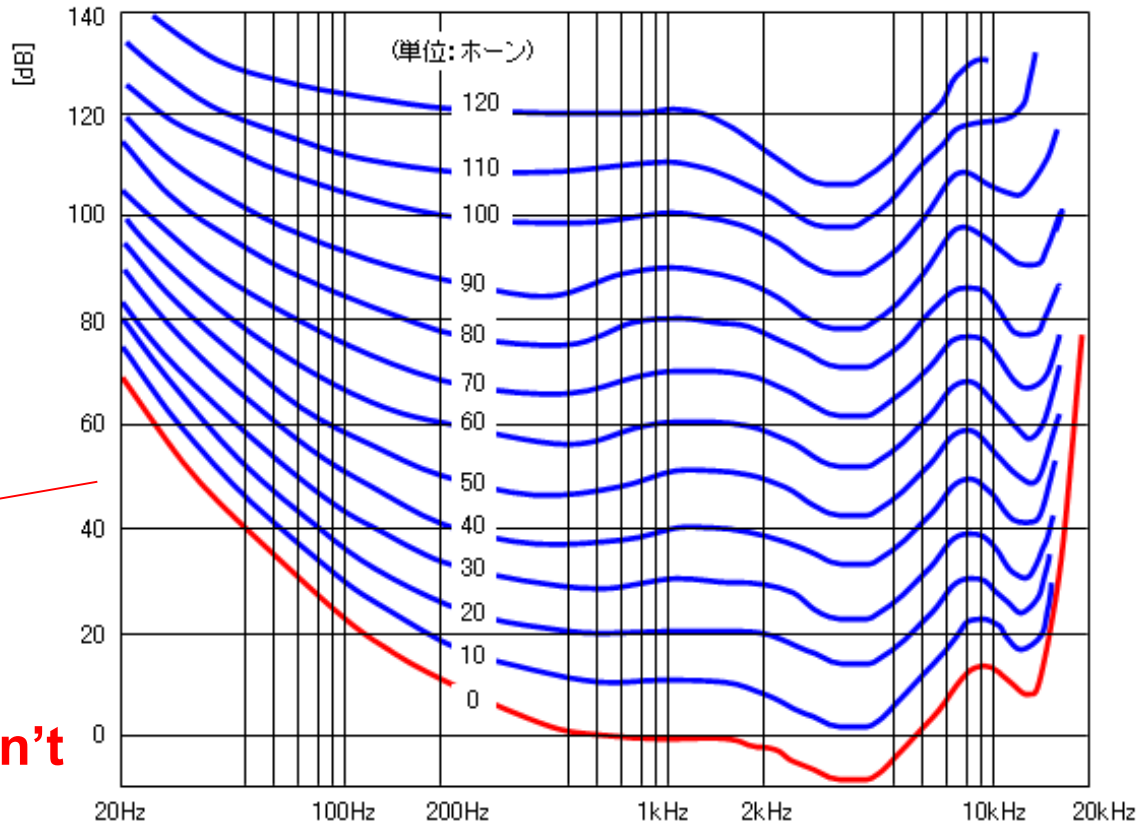
Provide short length code for the symbol which appear frequently. Use probability of appearance

ACBDAAACAAEBBAAB

	probability	symbol	code
High Probability	1/2	A	0
	1/4	B	10
	1/8	C	110
	1/16	D	1110
	1/16	E	1111

Basic technique for lossy audio coding (Introduction to psychoacoustics – 1)

Loudness contours from Fletcher and Munson



The hearing sensation that corresponds to sound levels is the loudness of the sound (Fig. right)

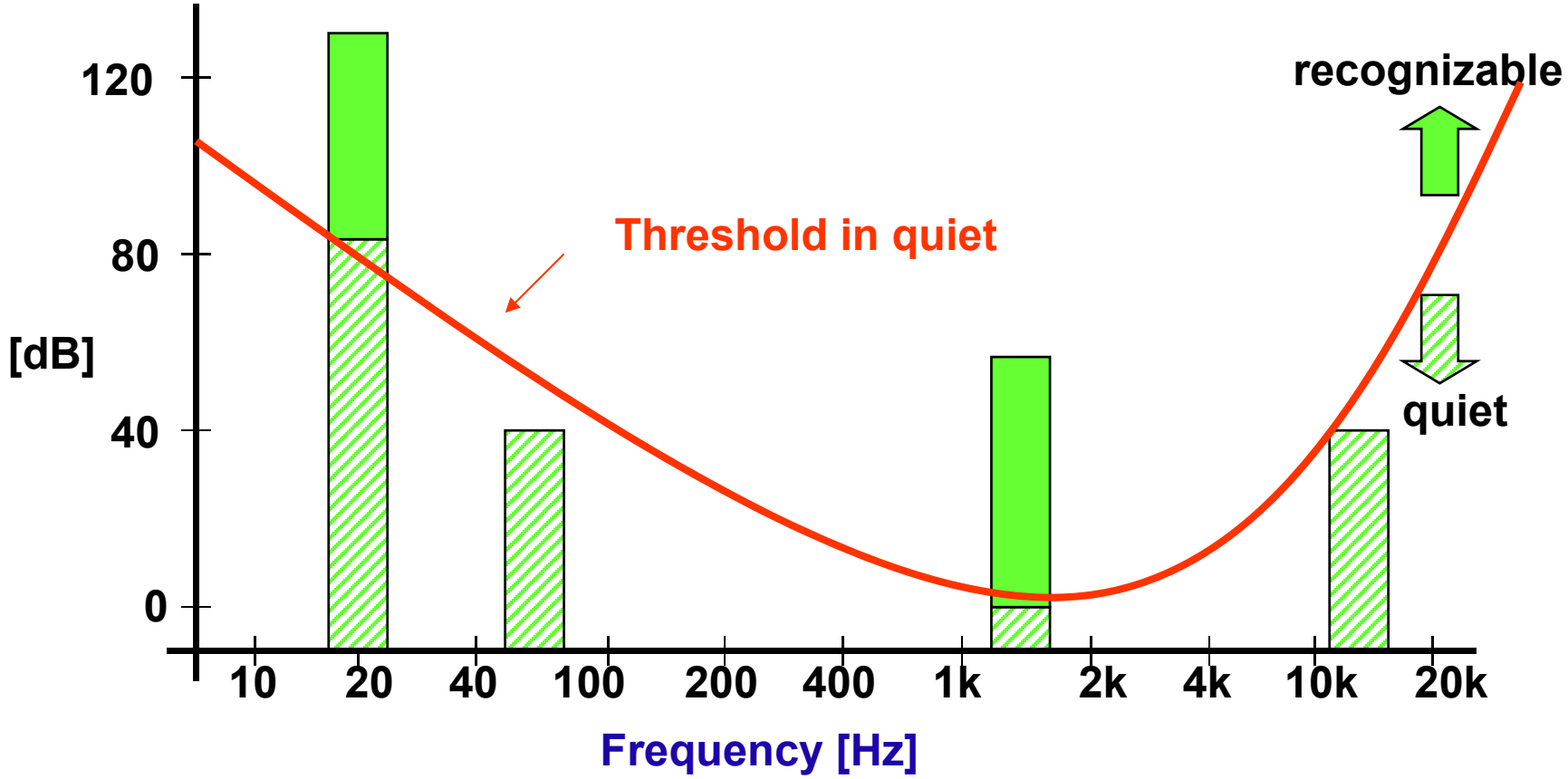
→Hearing range and hearing threshold are exist

ATH : Absolute Threshold Level

We can delete the information that we can't hear

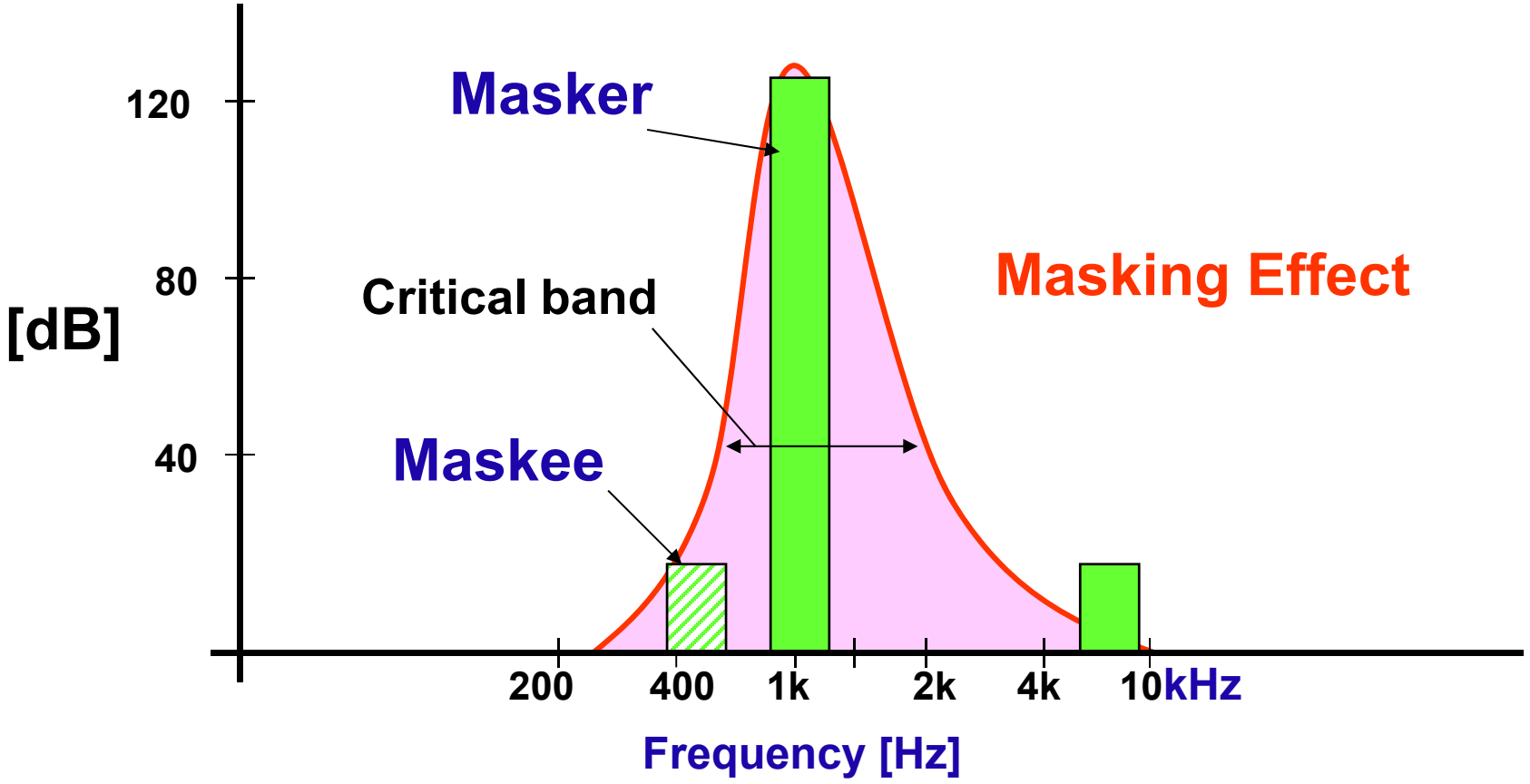
Basic technique for lossy audio coding (Introduction to psychoacoustics – 2a)

The sensitivity of the human being is not constant for the frequency of the signals.

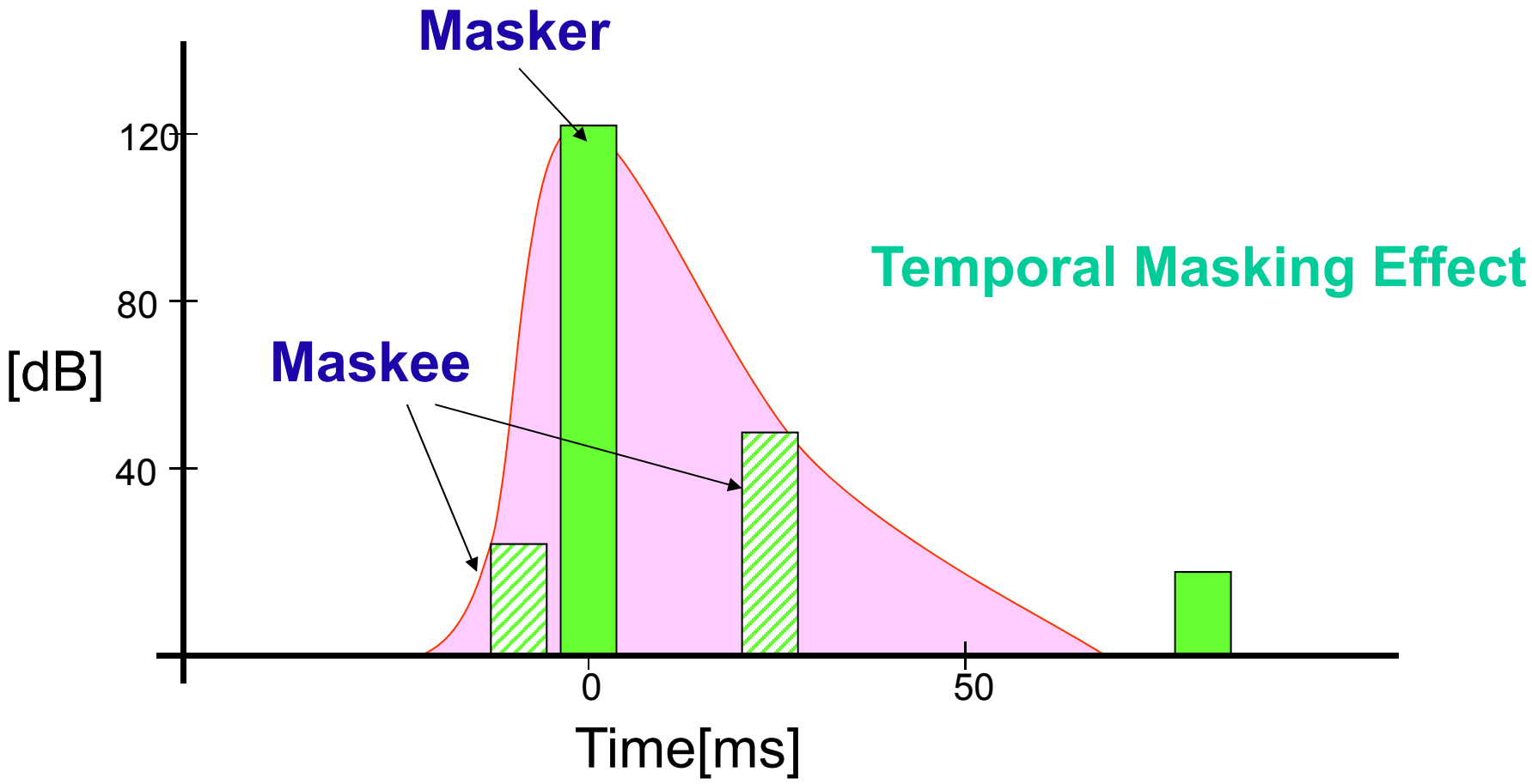


Basic technique for lossy audio coding (Introduction to psychoacoustics – 2b)

When a big sound exists, there is “masking” effect around the that sound.

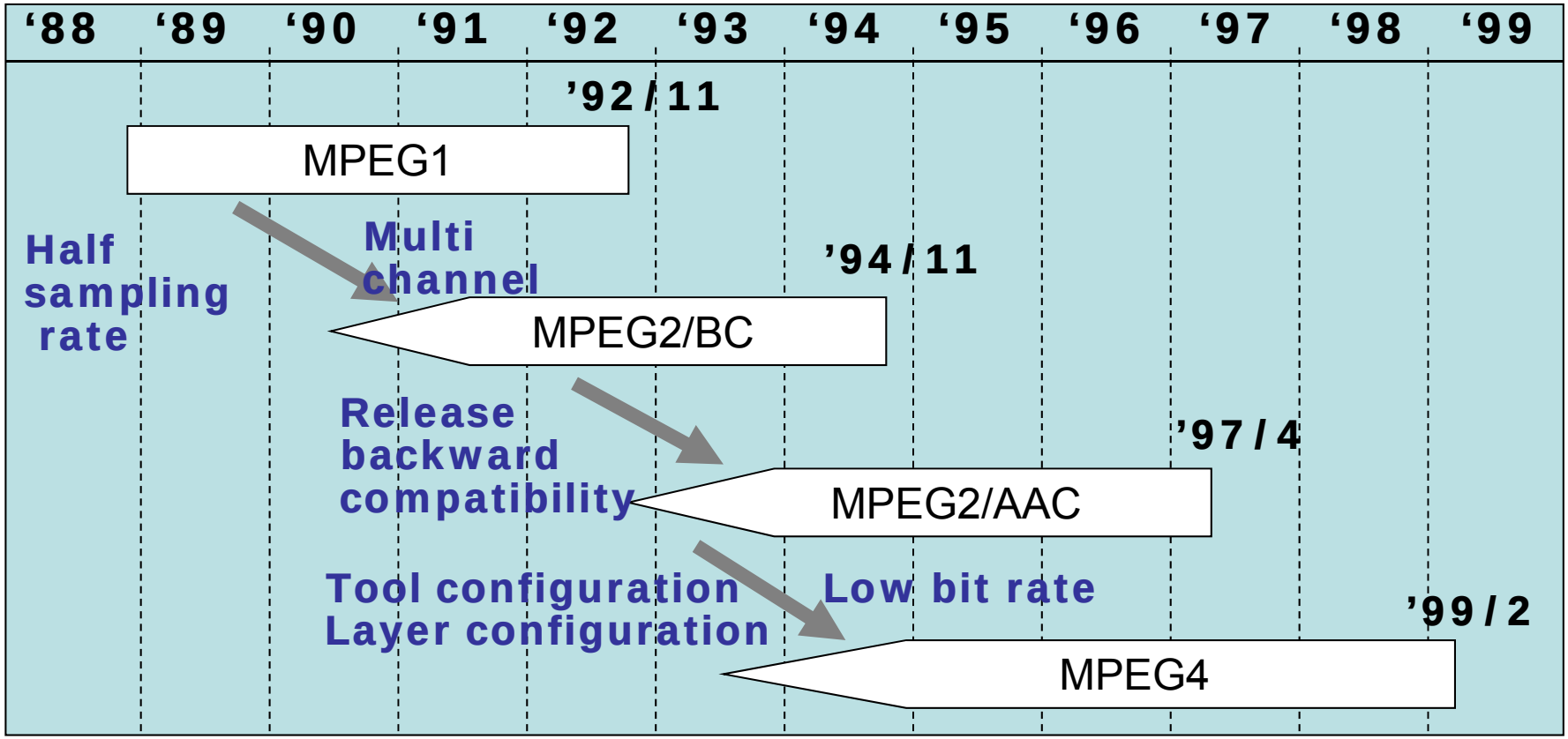


Basic technique for lossy audio coding (Introduction to psychoacoustics – 2c)



MPEG Audio Standard

The history of MPEG Audio standards is shown in the figure below. Standardization started from MPEG1 until MPEG4 as follows.



MPEG Audio Standard

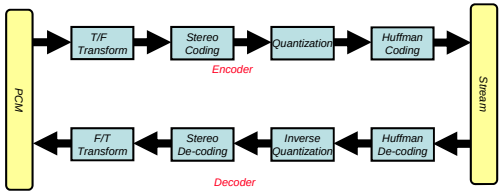
MPEG Audio and MPEG Audio/BC coding are standardized by ISO
ISO/IEC 11172(MPEG1) or 13818(MPEG2)

Basic features are as follows

Layer	Layer 1	Layer 2	Layer 3(MP3)
Mapping	Sub Band mapping	Sub band Mapping	Hybrid (sub band+DCT)
Bit Rate	32 to 448Kbps/ch	32 to 384kbps/ch	32 to 320kbps/ch
Target Bit Rate	128 to 192Kbps/ch	96 to 128kbps/ch	64 to 128kbps/ch
Sampling Frequency	48/44.1/32 KHz (and 24/22.05/16 KHz for MPEG2 half rate)		
Modes	Stereo/Joint Stereo/Dual/Single (and Multi channel up to 7.1ch for MPEG2)		

Note : Multiple stream can be multiplexed to make one stream.
Layer 1 & 2are based on “MUSICAM”.

Layer I & II processing flow



Input

Sub-band Analysis filter

Quantization (linear)

Bit compress

Build bit stream, generate CRC

Scale factor Selection info.

Scale factor extract

Side info. encode

FFT

Psych-acoustic model

Dynamic bit allocation

Encoder process

Storage Media channel

Output

Sub-band Synthesis filter

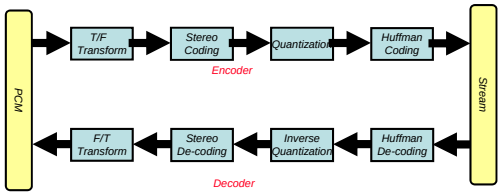
Inverse Quantization

Bit stream parser CRC check

Side info. decode

Decoder process

Dolby digital (AC-3) processing flow



Input
MDCT
Mantissa
Block length
Quantization
Psych-acoustic analysis
-
Pseudo Psych-acoustic
Differential encode
Multiplex Coupling
Exponent
Encoder process

Storage Media channel

Output
IMDCT
Mantissa
Block length
Inverse Quantization
+
Pseudo Psych-acoustic
Differential decode
DeMux Decupling
Exponent
Decoder process

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