Multimedia Communications

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Multimedia communications is the field referring to the representation, storage, retrieval and dissemination of information expressed in multiple media, such as text, voice, graphics, images, audio and video.

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Digital Image and Video

- With rapid computer technology advance, digital image processing gets a high attention:
- Stereoscope for 3-D robot vision
- Image Enhancement
- Image Restoration
- Image data compression
- Segmentation
- Image Description
- Image Recognition
- Document protection by watermarking

Image Enhancement

- The process of improving the quality of a digitally stored image by manipulating the image with software.

For example, to make an image lighter or darker, or to increase or decrease contrast.

Image Enhancement
Contrast Before

- 

Image Enhancement

After Contarst Enhancement

Image Enhancement

Image Enhancement

Brightness after Enhancement

Noisy Image Before

After Noise Reduction

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Medical Image Noise removal

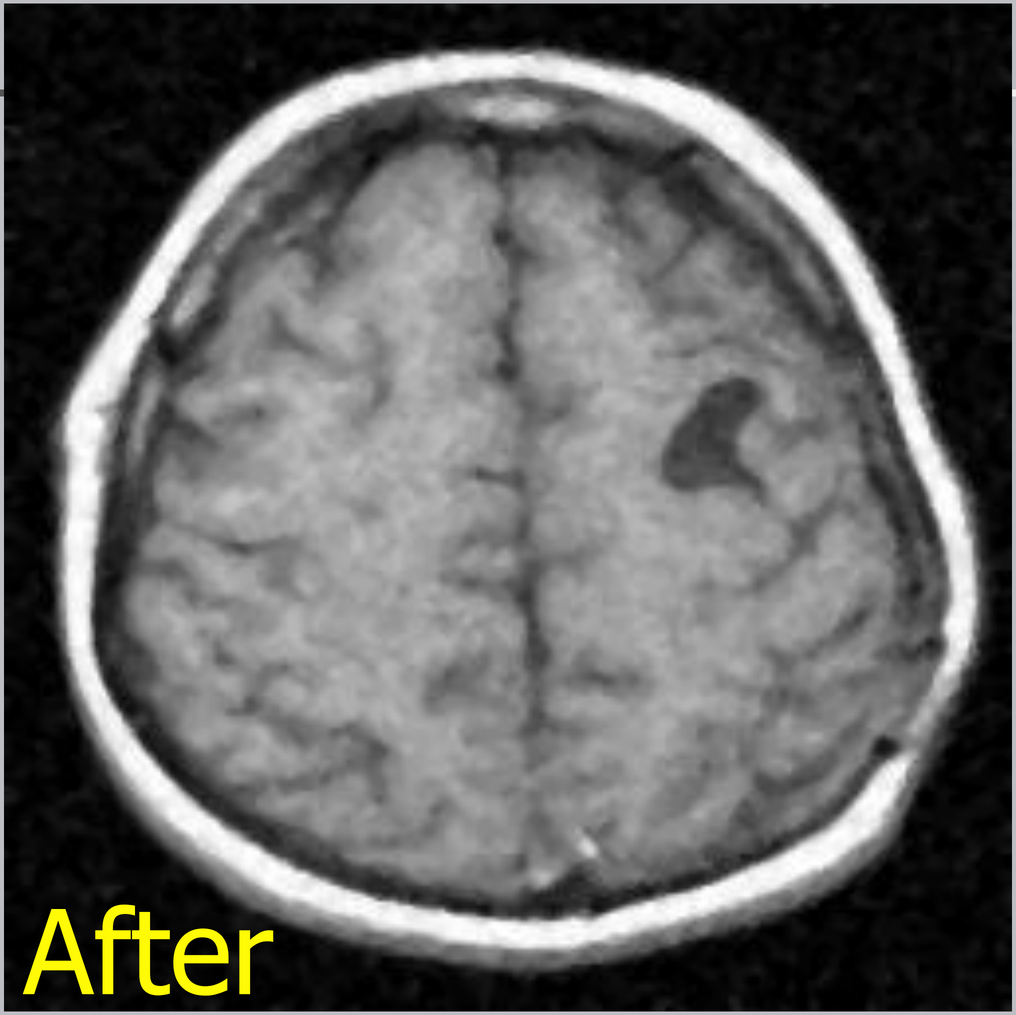

Image Restoration

- **In many applications (e.g., satellite imaging, medical imaging, astronomical imaging, poor-quality family portraits) the imaging system introduces a slight distortion.**

Blurred Image

Image Deblurring

Cracked Image

Restored Image

Image Data Compression

- Image data occupies a huge area of storage or transmission channel when it is used without compression.

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Discrete wavelet transform

- Like DCT, discrete wavelet transform mathematically transforms an image into frequency components. The process is performed on the entire image, which differs from the other methods (DCT), that work on smaller pieces of the desired data. The result is a hierarchical representation of an image, where each layer represents a frequency band.

Discrete wavelet transform
Common Applications

- Lossy data compression
- De-noising
- Detection

Compression standards

- MPEG stands for the Moving Picture Experts Group. There are five MPEG standards being used or in development. Each compression standard was designed with a specific application and bit rate in mind, although MPEG compression scales well with increased bit rates. They include:

MPEG-1

- Designed for up to 1.5 Mbit/sec Standard for the compression of moving pictures and audio. This was based on CD-ROM video applications, and is a popular standard for video on the Internet, transmitted as .mpg files. In addition, level 3 of MPEG-1 is the most popular standard for digital compression of audio--known as MP3. MPEG-1 is the standard of compression for VideoCD

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MPEG-2

- Designed for between 1.5 and 15 Mbit/sec Standard on which Digital Television set top boxes and DVD compression is based. It is based on MPEG-1, but designed for the compression and transmission of digital broadcast television. MPEG-2 scales well to HDTV resolution and bit rates, obviating the need for an MPEG-3.

MPEG-4

- Standard for multimedia and Web compression. MPEG-4 is based on object-based compression, similar in nature to the Virtual Reality Modeling Language. Individual objects within a scene are tracked separately and compressed together to create an MPEG4 file. This results in very efficient compression that is very scalable, from low bit rates to very high. It also allows developers to control objects independently in a scene, and therefore introduce interactivity.

MPEG-7

- This standard, currently under development, is also called the Multimedia Content Description Interface. When released, the group hopes the standard will provide a framework for multimedia content that will include information on content manipulation, filtering and personalization, as well as the integrity and security of the content. Contrary to the previous MPEG standards, which described actual content, MPEG-7 will represent information about the content.

MPEG-21

- work on this standard, also called the Multimedia Framework, has just begun. MPEG-21 will attempt to describe the elements needed to build an infrastructure for the delivery and consumption of multimedia content, and how they will relate to each other.

JPEG

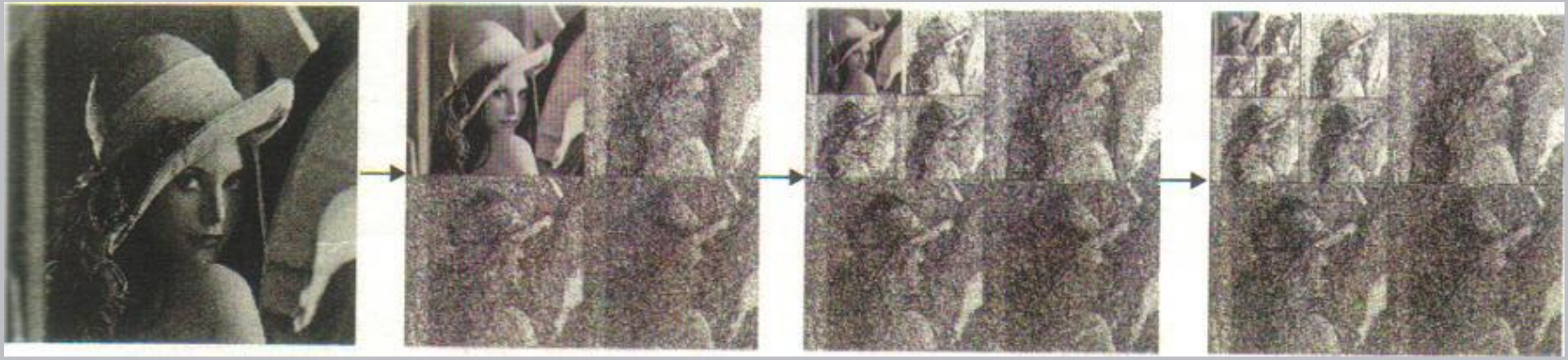
- JPEG stands for Joint Photographic Experts Group. It is also an ISO/IEC working group, but works to build standards for continuous tone image coding. JPEG is a lossy compression technique used for full-color or gray-scale images, by exploiting the fact that the human eye will not notice small color changes.

JPEG 2000

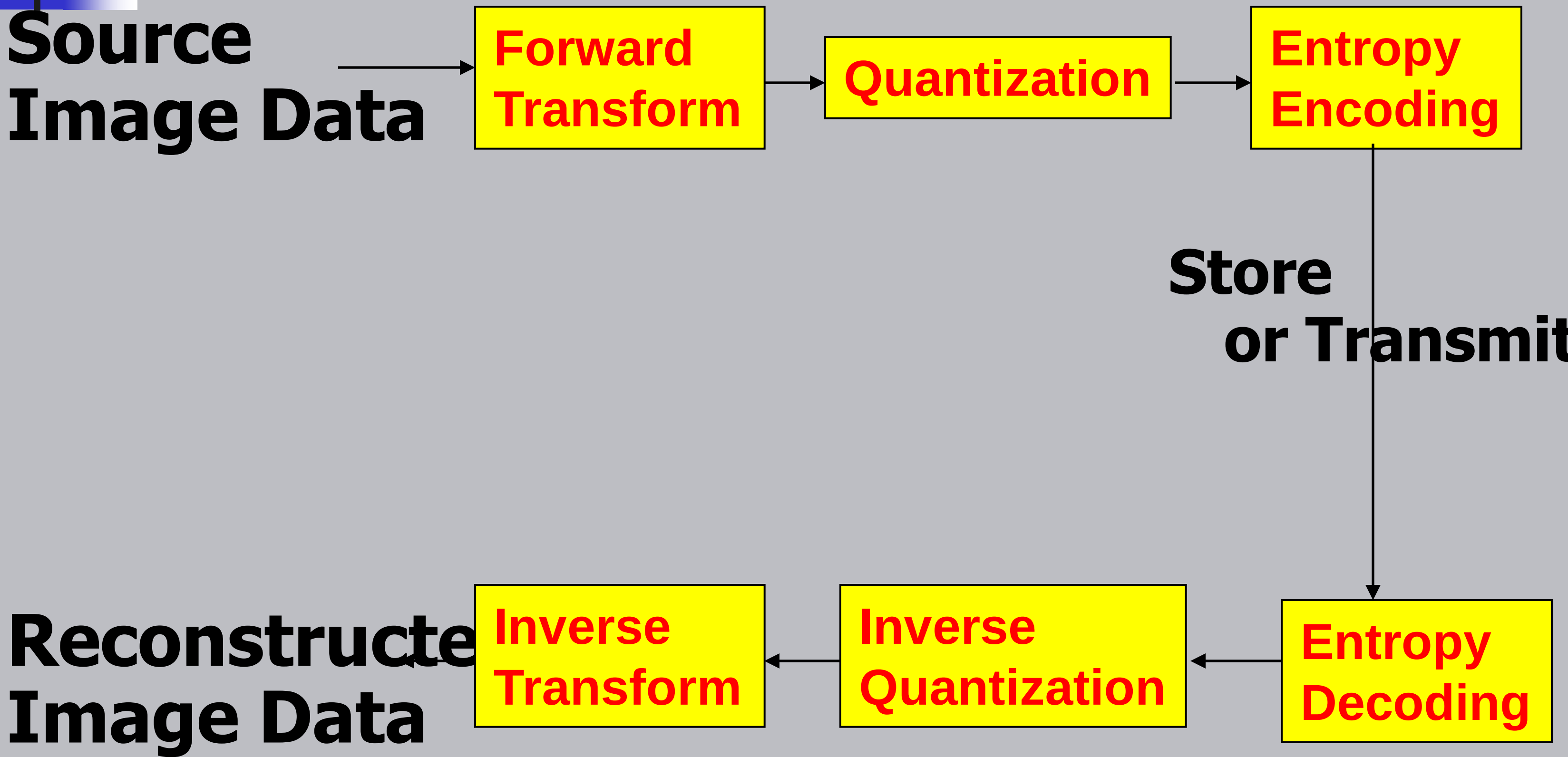
- JPEG 2000 is an initiative that will provide an image coding system using compression techniques based on the use of wavelet technology.

Wavelet Transform

- **Convolution based filtering**
- **Lifting based filtering**



General block diagram of the JPEG 2000



ROI (Region of interest)

Often there are parts of an image that are more important than others. This feature allows users to define certain ROI's in the image to be coded and transmitted with better quality and less distortion than the rest of the image.

Example of ROI mask

Example of JPEG2000

(a) (b)

Reconstruction image “ski” after compression at 0.25 b/p by means of

(a) JPEG (b) JPEG 2000.

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Image Segmentation

- Segmentation is to distinguish objects from background.

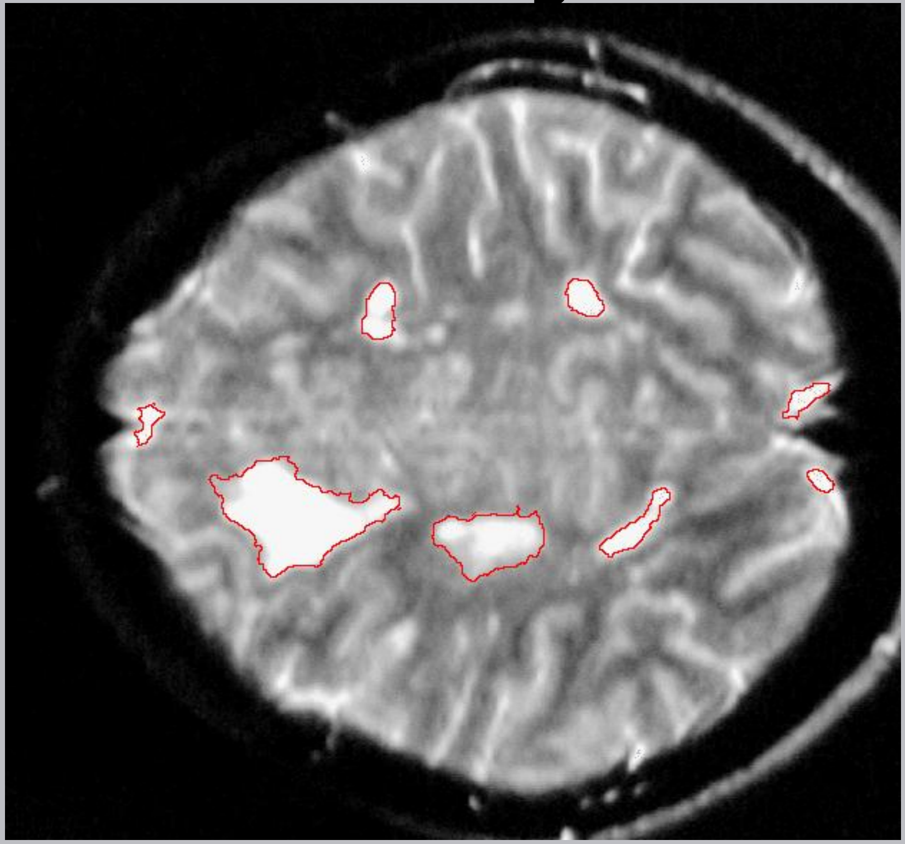


Image Segmentation

There are four techniques for image segmentation:

- Threshold techniques
- Edge-based methods
- Region-based techniques
- Connectivity-preserving relaxation methods

Threshold techniques

- Threshold techniques, which make decisions based on local pixel information, are effective when the intensity levels of the objects fall squarely outside the range of levels in the background. Because spatial information is ignored, blurred region boundaries can create havoc.

Edge-based methods

- Edge-based methods is based on contour detection: their weakness in connecting together broken contour lines make them, too, prone to failure in the presence of blurring.

Edge Detection Operator

(a)Prewitt･Operator

-1	0	1
-1	0	1
-1	0	1

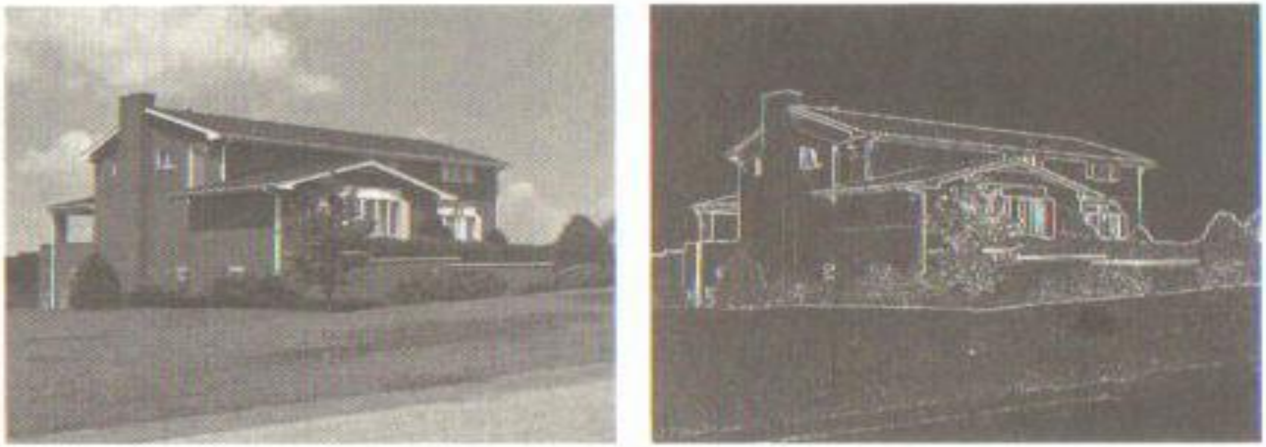
-1	-1	-1
0	0	0
1	1	1

(b) Sobel･Operator

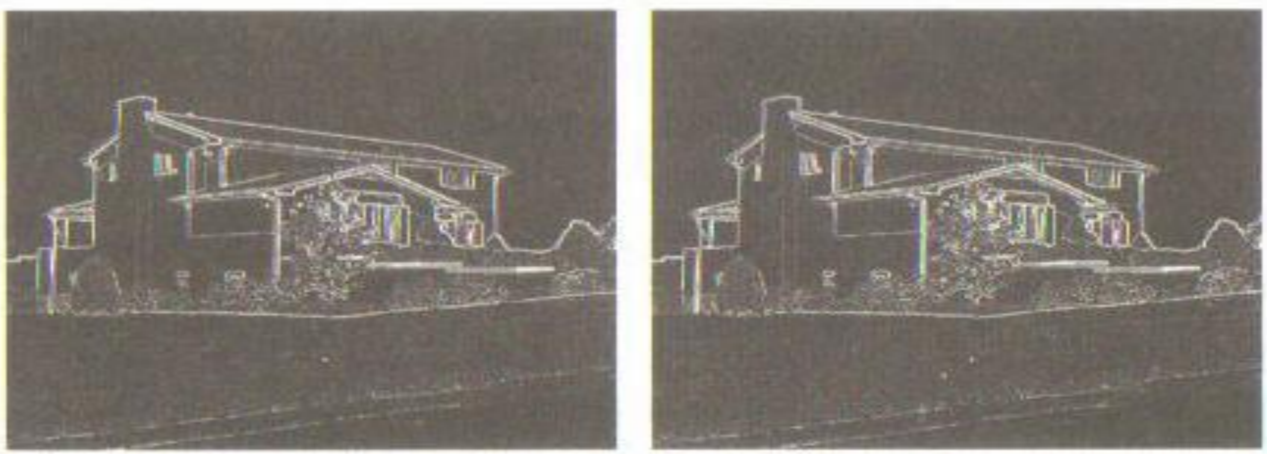
-1	0	1
-2	0	2
-1	0	1

-1	-2	-1
0	0	0
1	2	1

Derivative Images (Edges)



(a) 原画像　(b) ロバーツ・オペレータ



(c) プリューウィット・オペレータ　(d) ソーベル・オペレータ

Region-based method

- A region-based method usually proceeds as follows: the image is partitioned into connected regions by grouping neighboring pixels of similar intensity levels. Adjacent regions are then merged under some criterion involving perhaps homogeneity or sharpness of region boundaries.

Connectivity-preserving relaxation method

- The main idea in connectivity-preserving relaxation-based segmentation method, is to start with some initial boundary shape represented in the form of spline curves, and iteratively modify it by applying various shrink/expansion operations according to some energy function.

Document protection by watermarking

- digital watermark : A special message embedded in an image.

Original unwatermarked image

Digimarc watermarked image

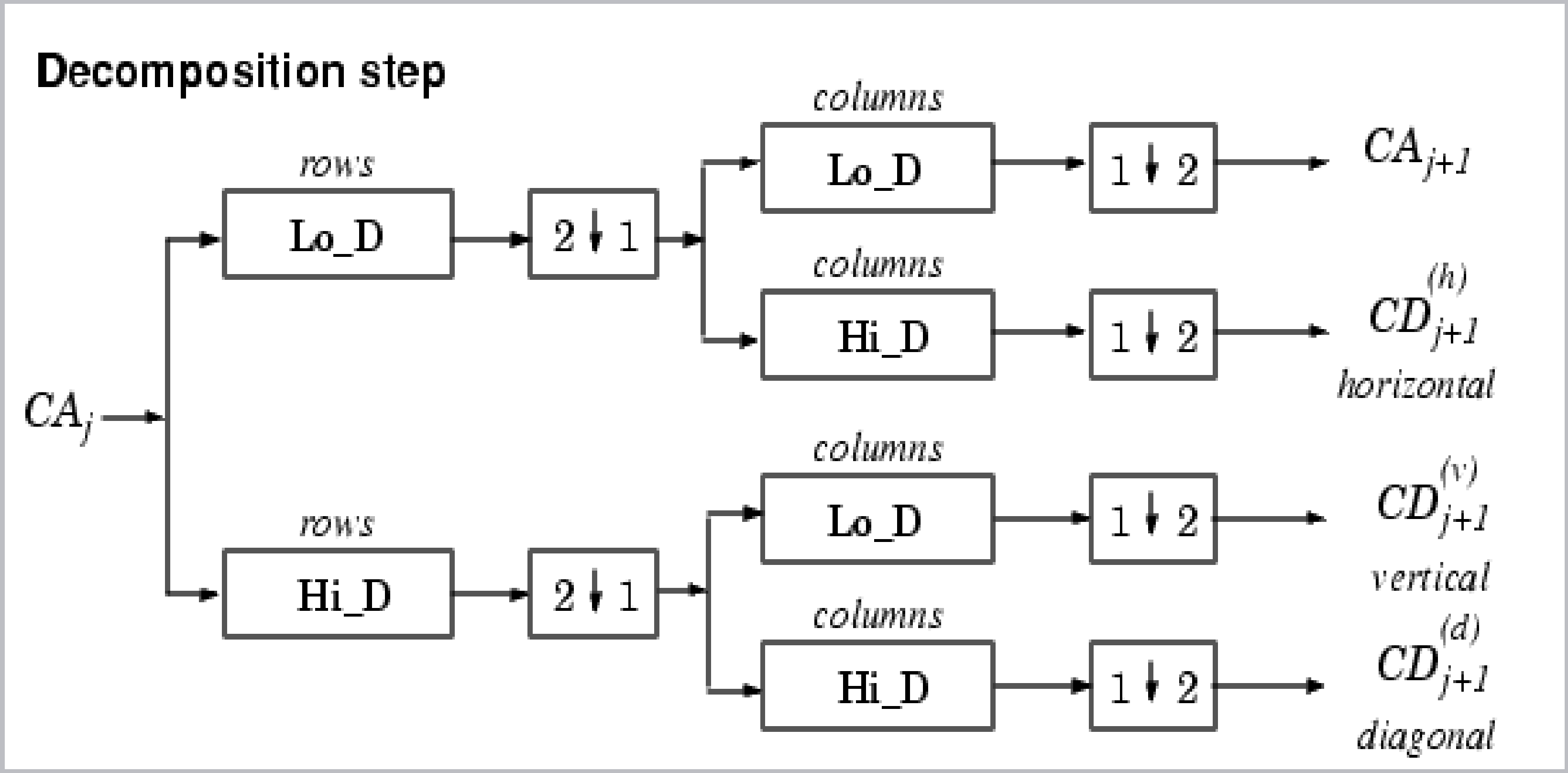
Exaggerated view of imperceptible Digimarc watermark

Restoration method for deteriorated images by Distortion and Noise

: Research of Image Fusion by using Wavelet Transformation

Wavelet Transform

- Resolve to Approximation and Detail.



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2-D wavelet transform

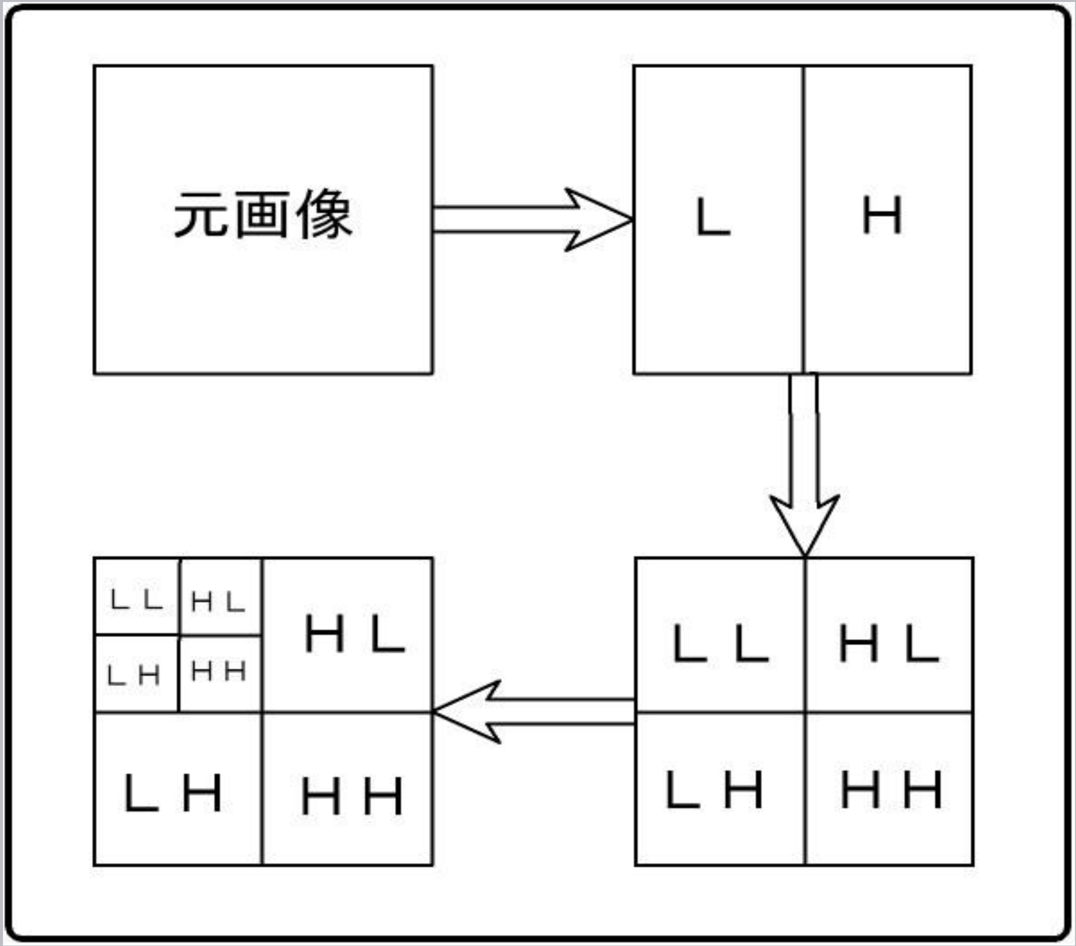
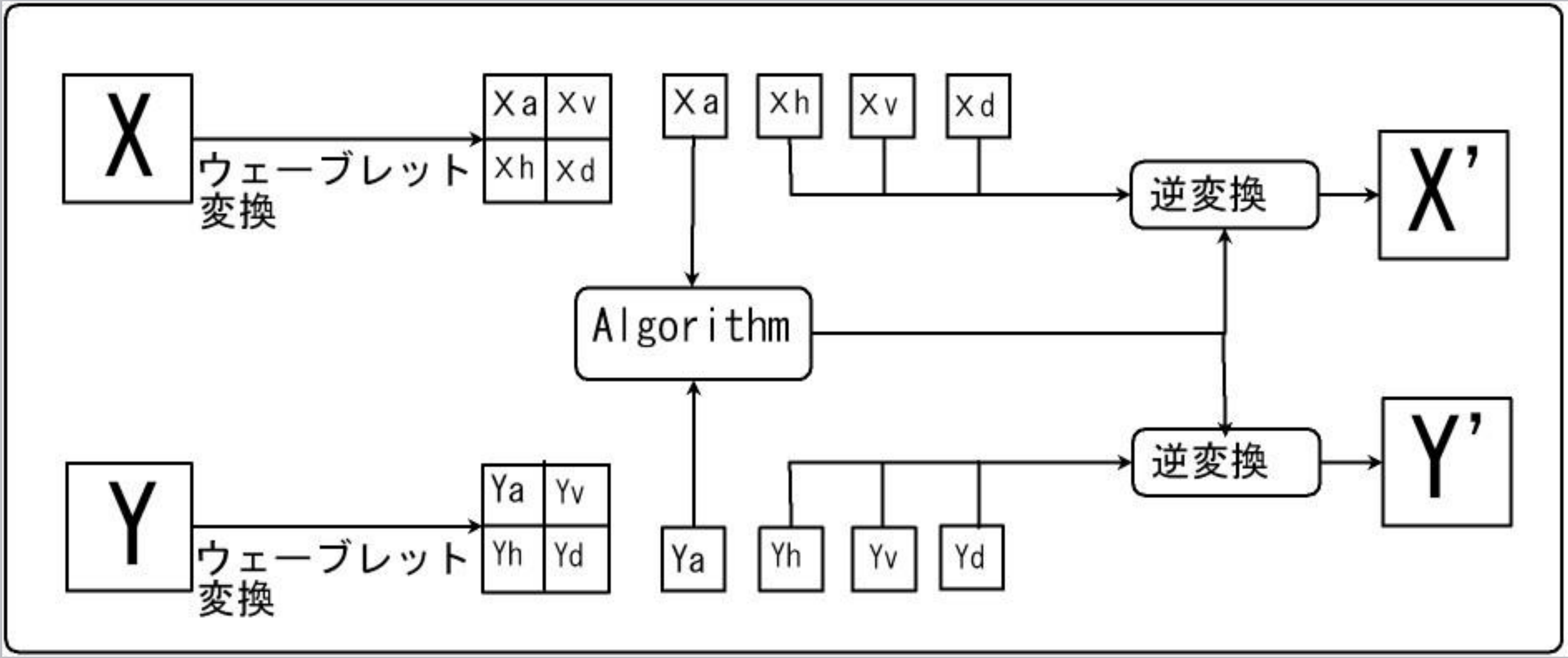


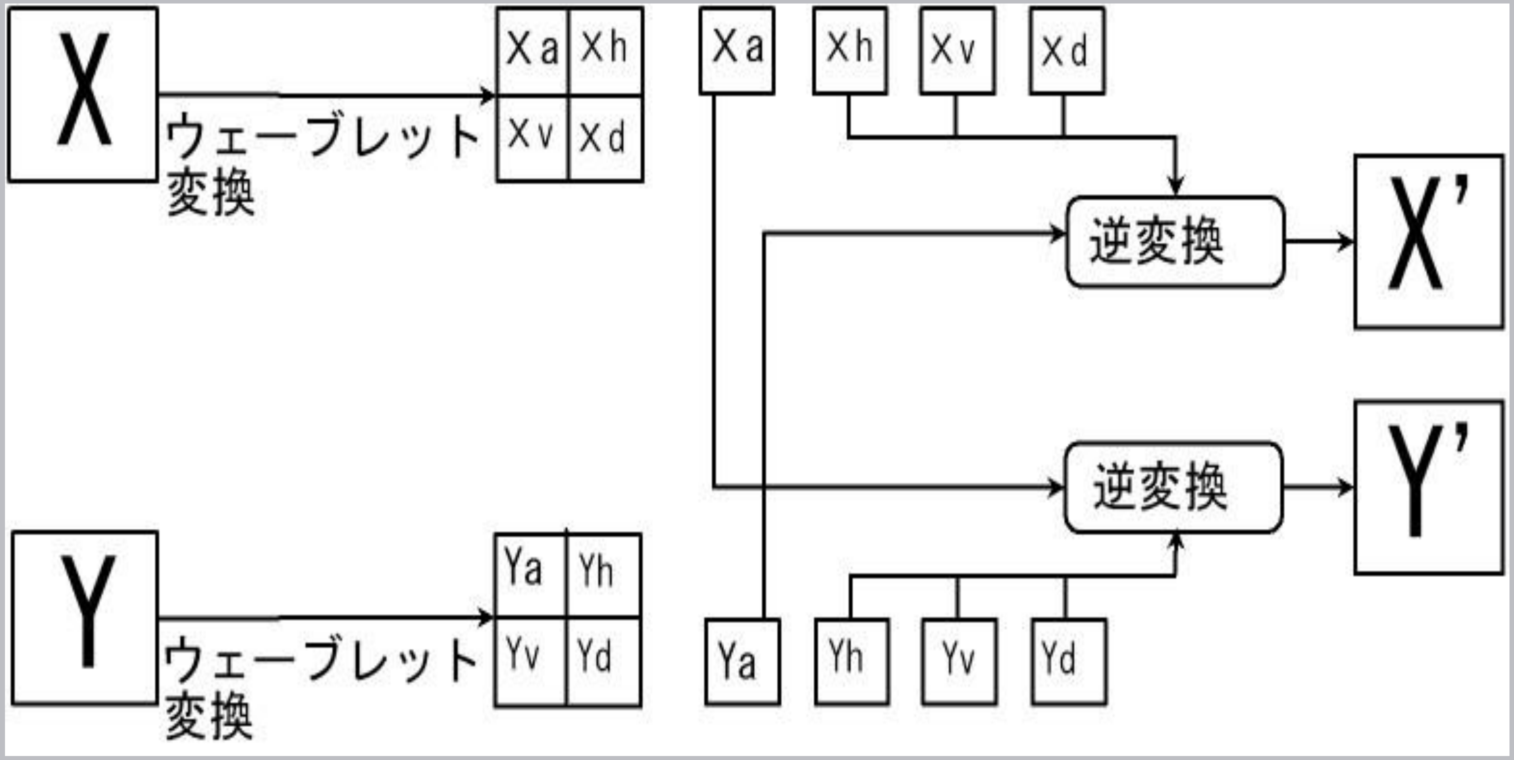
Image Fusion Algorithm

- RMS
- Mean



Replacement

- Replace Approximation and Detail.



Result : 1

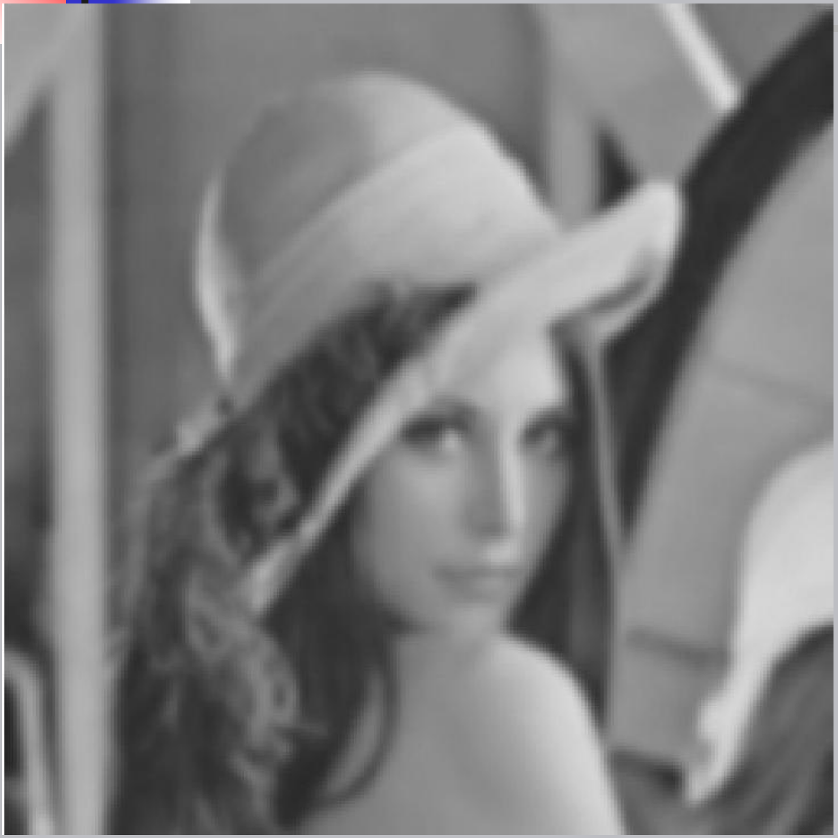


Fig1：distorted image

Fig2：Noisy image

Result : 2

Fig3 : RMS

Fig4 : Mean

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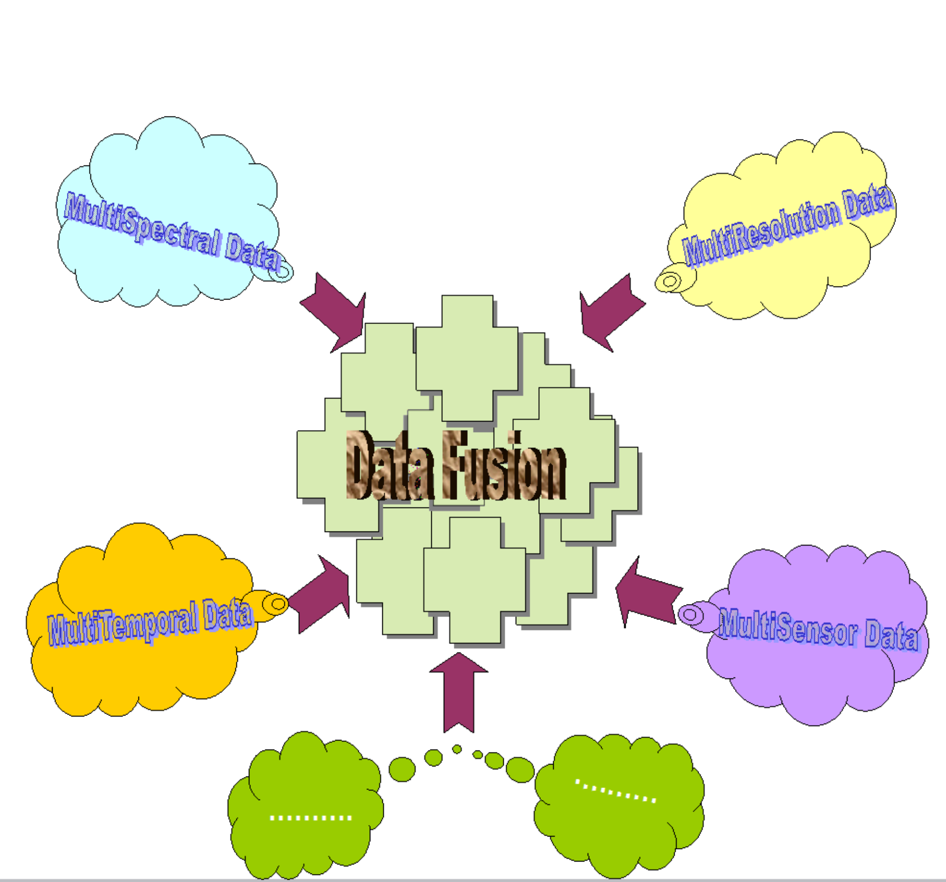
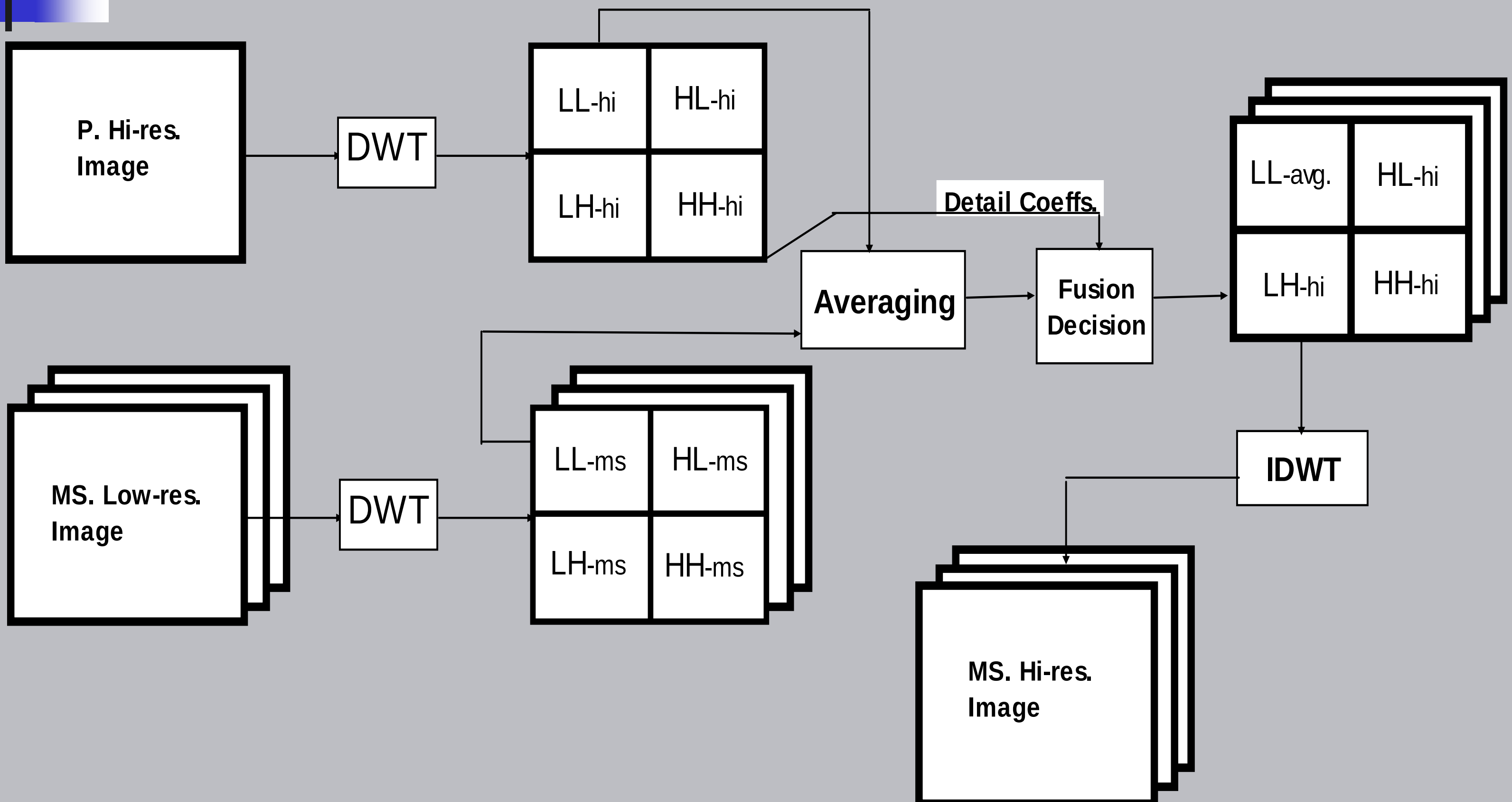
Data Fusion



- Data fusion deals with the combination of information made available by various knowledge sources such as sensors, in order to provide a better understanding of a given scene.
- Fusion of digital image data becomes a valuable tool in **Remote Sensing** images.

Image Data Fusion



Flowchart of wavelet-based image fusion procedure

High Spectral Resolution

High Spatial Resolution

High Spectral & Spatial Resolution

Back

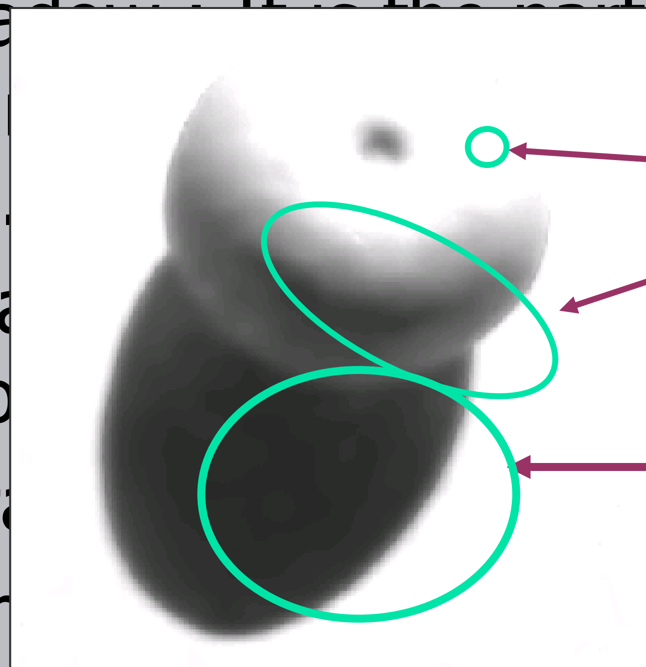
Homomorphic Processing Approach for Image Shadow Identification

Type of Shadows

- Self Sha … of shadow on the mai … ot illuminated by light.
- Cast sha … ect's shadow on backgro … d to two parts:

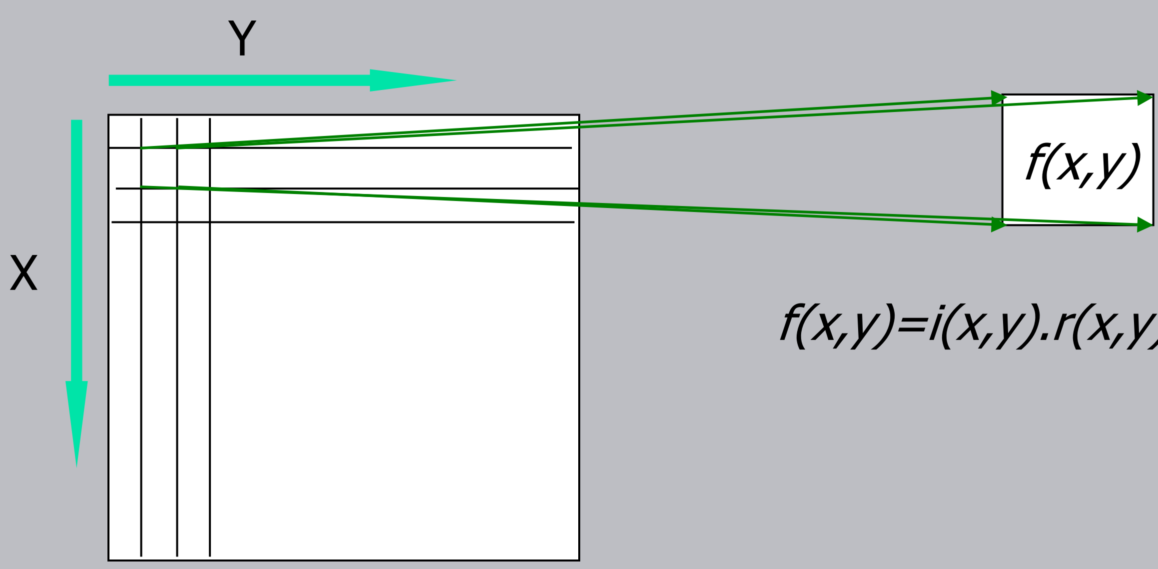
1- Umbr…

2- Penur…



Proposed Approach

- In the algorithm proposed in this study homomorphic approach is implemented



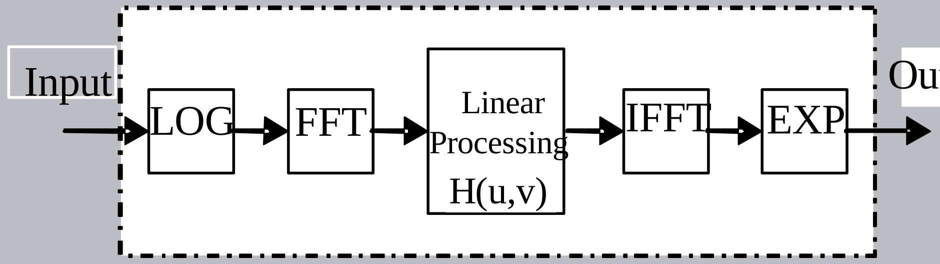
$$f(x,y)=i(x,y).r(x,y)$$

Homomorphic System

- $0 \le f(x,y)=i(x,y).r(x,y) < \infty$
- *f(x,y):* gray-level of image pixel which has two components
- *i(x,y):* illumination component which is non-zero and finite
- *r(x,y):* reflection component which is between 0(total absorption) and 1(total reflection)

Homomorphic System

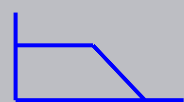
- In the first process of the Homomorphic system, two components of gray-level will converted to addition by taking logarithm in order to separately filter operation on illumination and reflection in Frequency Domain.



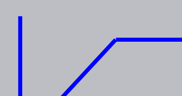
Homomorphic System

Shadow Identification

- The illumination component of an image generally is characterized by slow spatial variation.



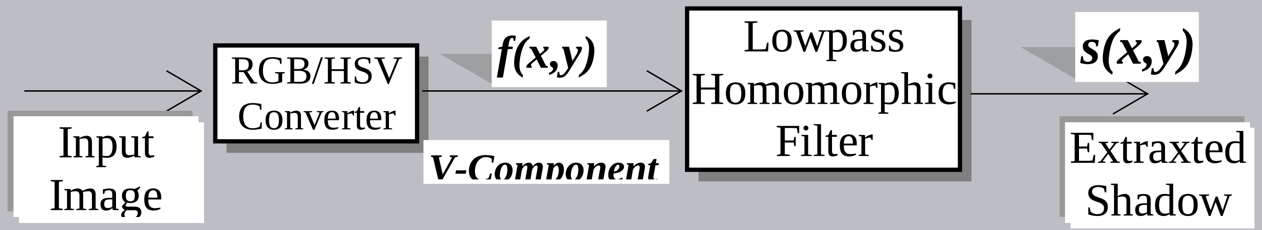
- The reflection component tends to vary abruptly, particularly at the junctions of dissimilar objects.



- These characteristics lead to associate the low frequency components of the Fourier transform of an image with the illumination and the high frequencies with the reflection.

Shadow Identification

- In the second process, by using appropriate LPF we will emphasize on illumination changes which is segmented as shadow area.



Shadow Identification

- In the third process, after background equalization, subtracting two filtered and equalized images in order to identify the shadow area.

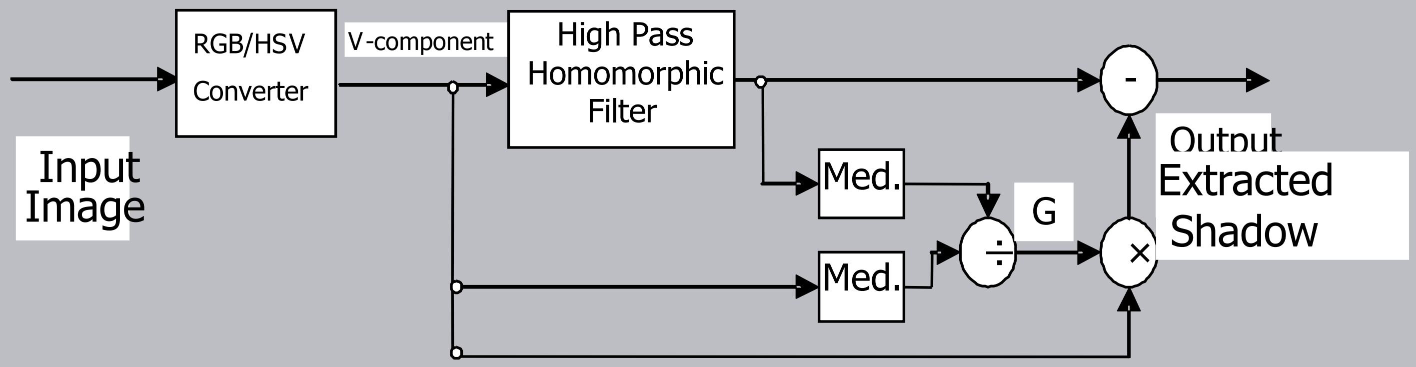
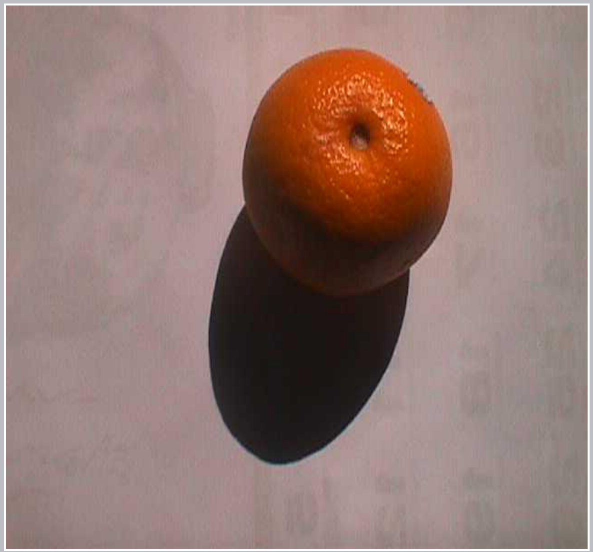
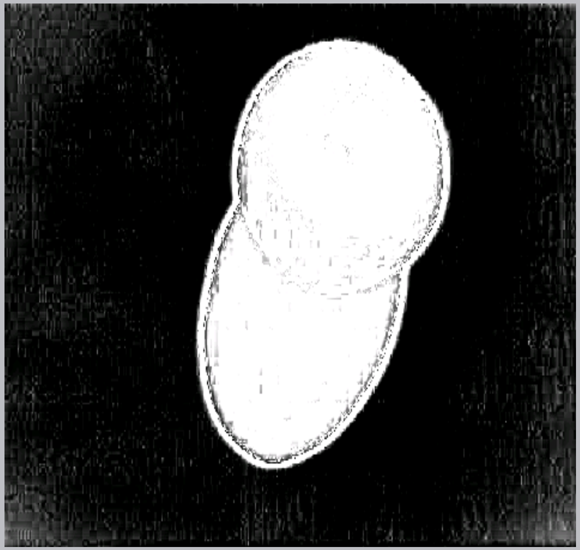


Fig. 6. Block diagram of Proposed System

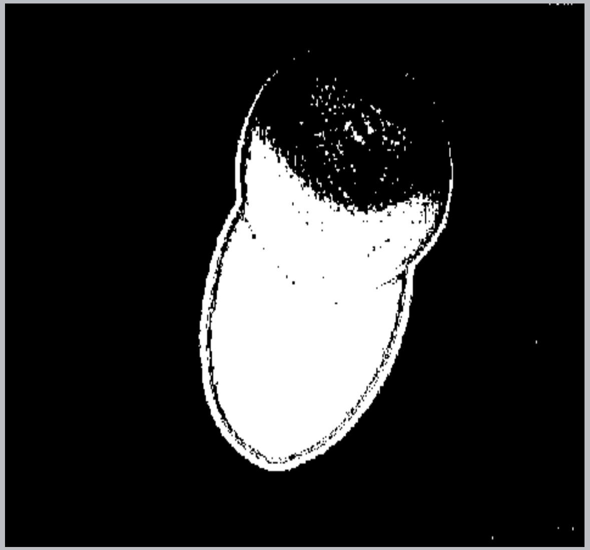
Experimental test results



RGB color space

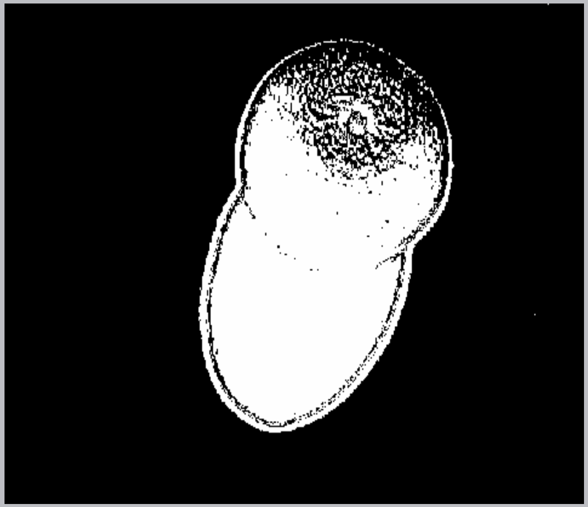


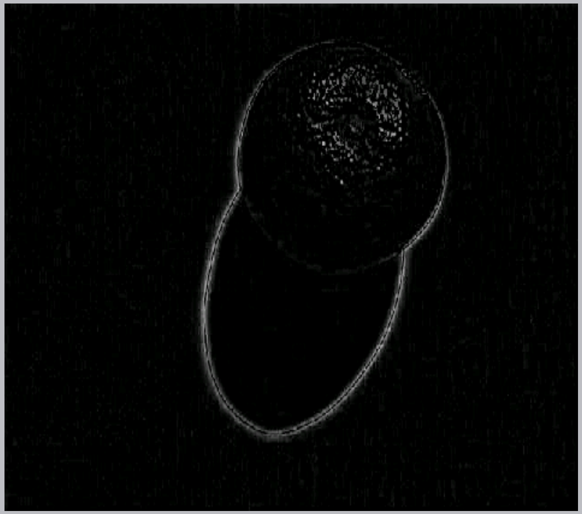
Using approach by Blue component



Using approach by Red (Dominant) component

Experimental test results





The result of approach using gray-scale image

Result of processing without using Homomorphic approach

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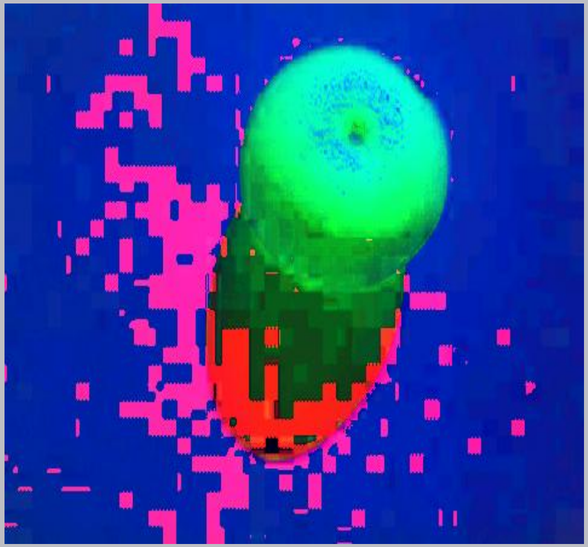
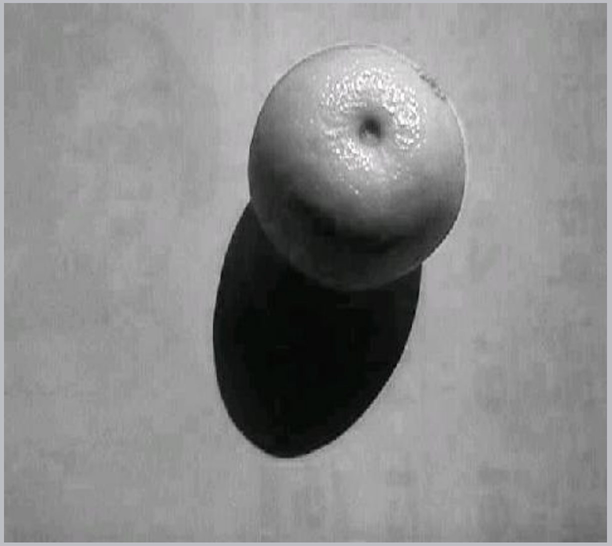
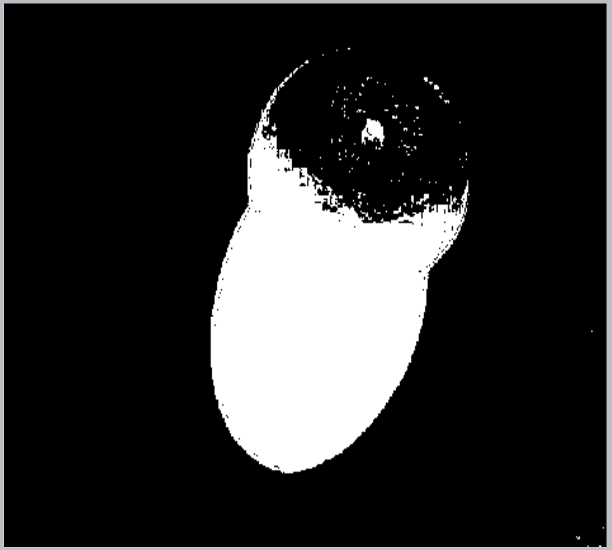
Experimental test results



HSV color space



V Component



Shadow detected

Applications

- Cloud shadow identification and data visibility in cloud shadow covered area in Remote Sensing images.
- Shadow detection using in Mobile Robotic Vision to identify the object from its shadow specially in the unknown area (Moon surface).
- Shadow detection in Arts and Painting.
- Shadow detection in moving object to identify the real object in control traffic system and etc.

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Image Cleaning by Median Filtering

WOMAN【noise → Salt & Pepper】
SNR : 15dB

Noisy Image

Manual Median filter for Noise Canceling

Moving average / Alpha-trimmed mean filter

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Alpha−trimmed mean filter

Midpoint filter / Edge Preserved Smoothing

Midpoint filter

Edge Preserved Smoothing

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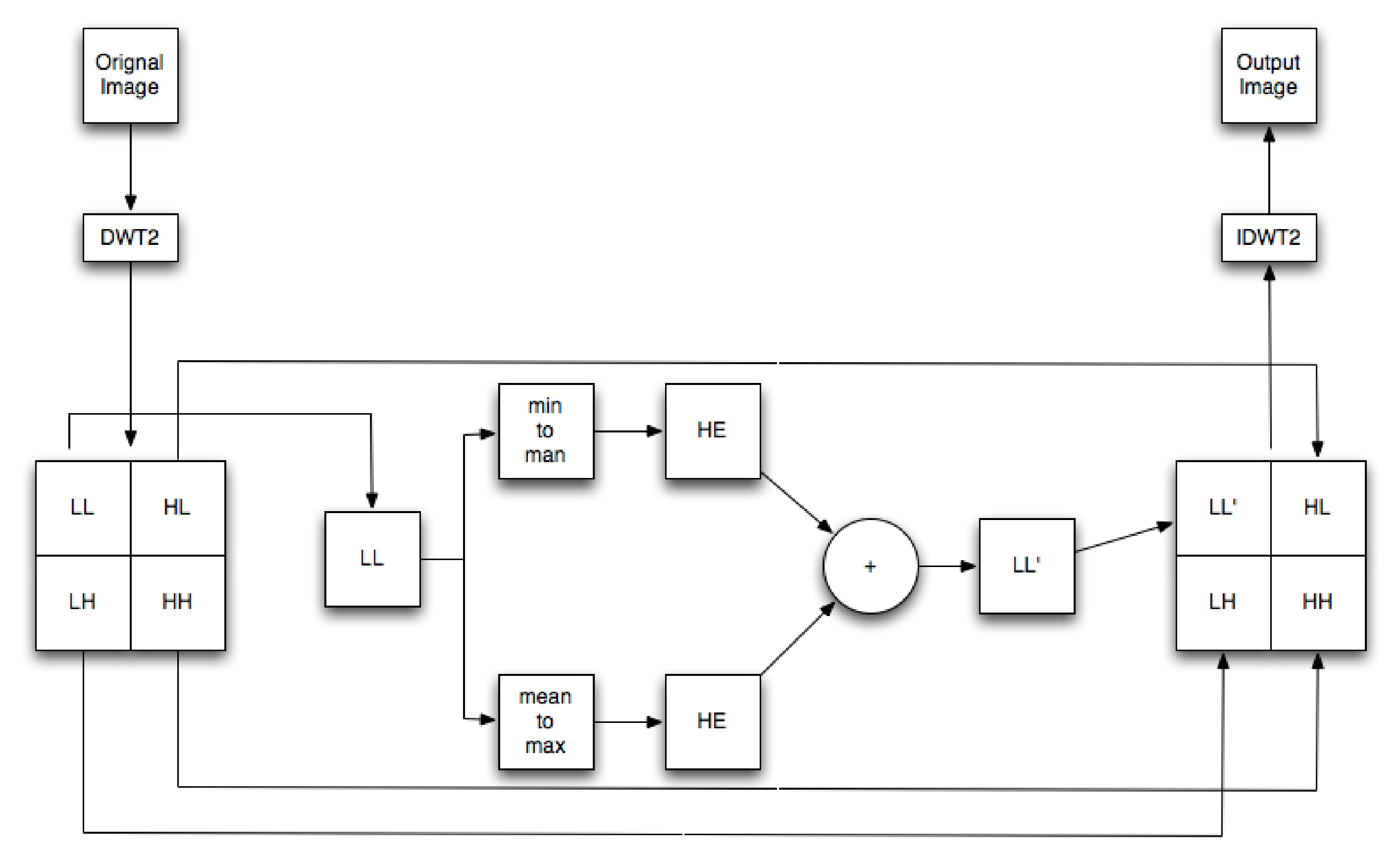
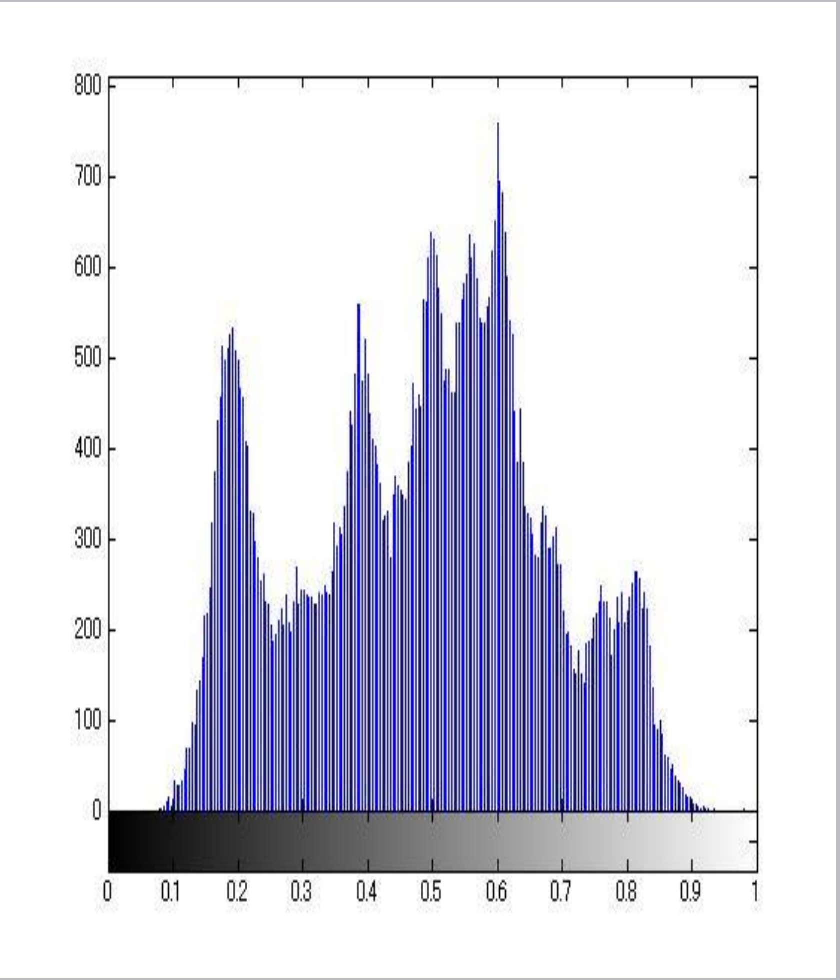
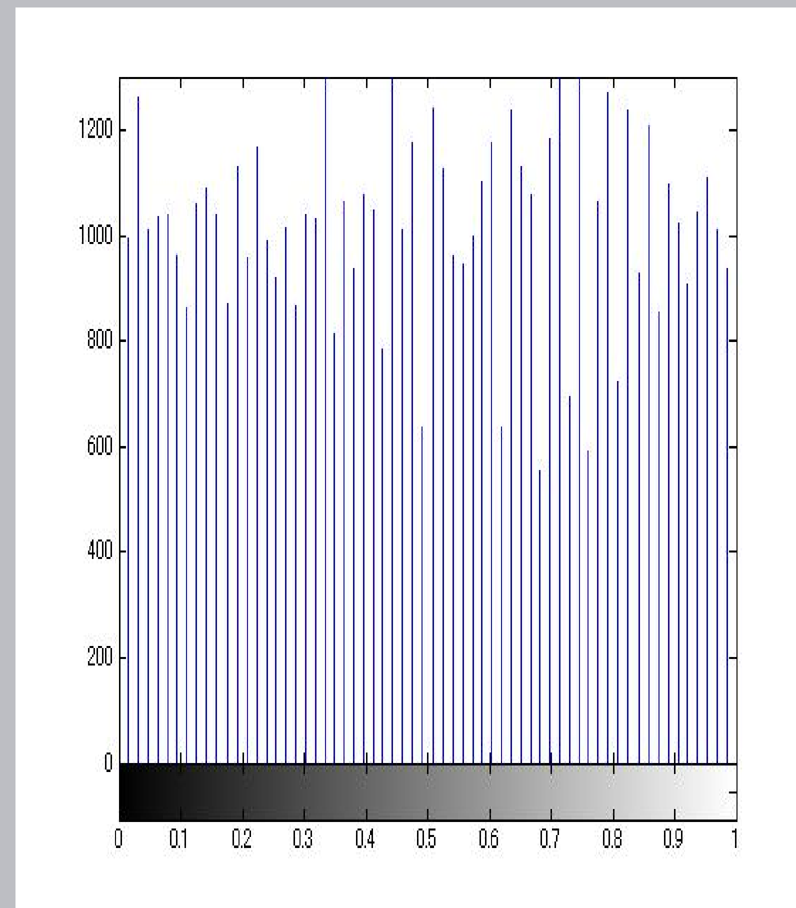
Image Enhancement Using Wavelet LL Histogram Separation



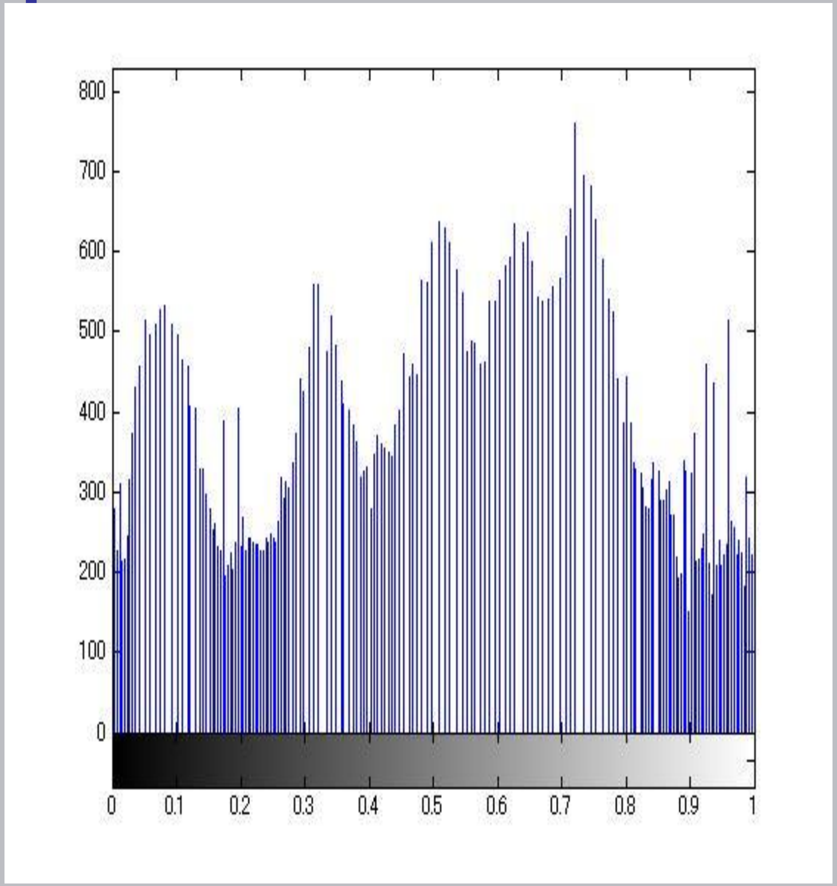
Original Image & Histogram



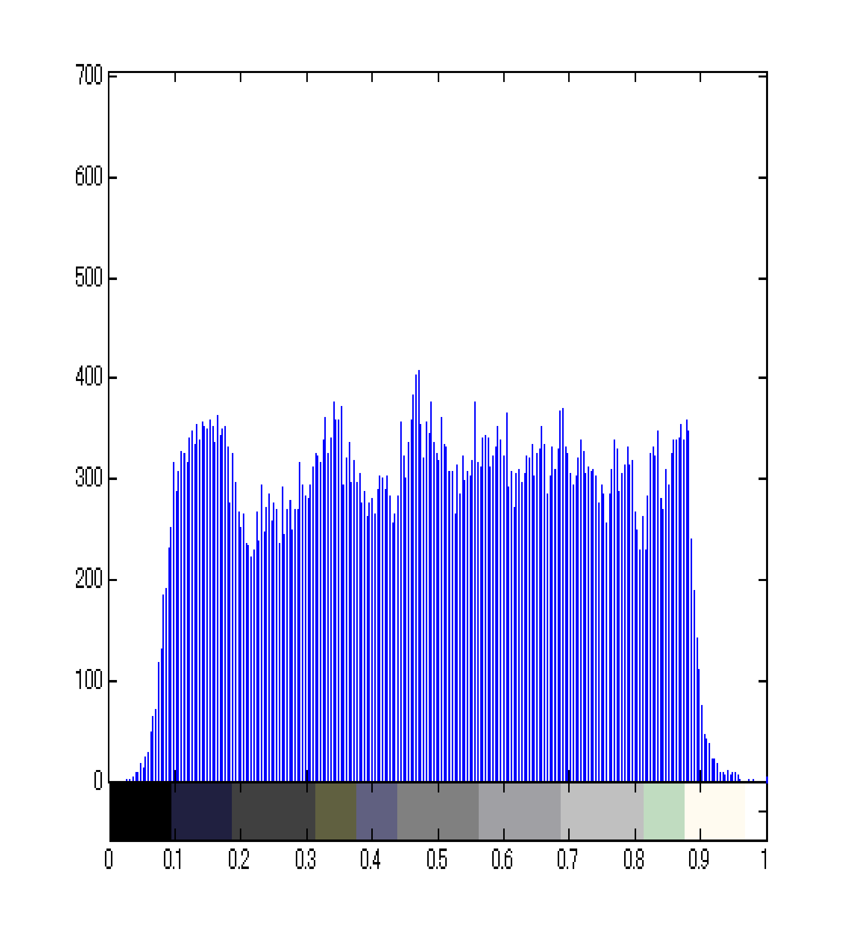
Histogram Equalized Image



Mean Value Histogram Separated Equalization

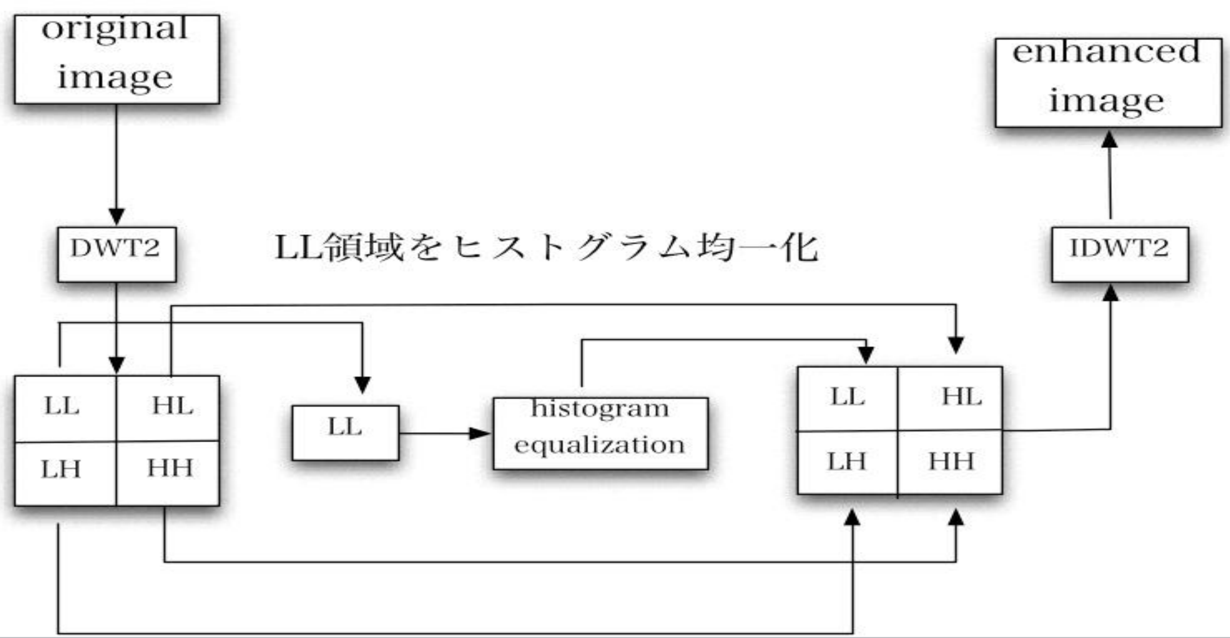


Wavelet LL Histogram Separation



Histogram Equalization in Wavelet Domain (LL)

LL領域のヒ ストグラム均一化



Comparison between LL,HH of Wavelet and Histogram Equalization

元画像 EME=0.8483

(LL)ヒストグラム均一化後の画像 EME=1.8654

(HH)ヒストグラム均一化後の画像
EME=1.8638

ヒストグラム均一化後の画像(HE) EME=1.8637

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Image Deblurring

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Frontal image

Skin detection using conventional method

Using proposed method

Automatic Facial Skin Detection Based on Gaussian Mixture Model Under Varying Illumination

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Image Enhancement Using Logarithm Bi-Histogram

Original
Pentagon
EME=25.03

Enhanced
image (HE)
EME=41.52

Enhanced
image (TBHE)
EME=66.81

TOP

Image Enhancement Using Splitting α-Rooting Method in Wavelet Domain

Original image EME=16

HE EME=23

S α-rooting EME=30

WS α-rooting EME=31

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Speech Coding

- Speech coding is a technique sometimes referred to as lossy coding.
- The input and output signals are not mathematically equivalent but they are perceptually similar.
- Differences can be heard, but are hopefully not annoying or are acceptable for the application. Traditionally speech coding is used for communication applications using telephony bandwidth speech (200 Hz - 3.5 kHz).
- However, changes in the communication infra-structure have opened the door for new exciting algorithms targeting all types of bandwidths from 3.5 kHz all the way up to CD quality sound.

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Speech Coders

Designing speech coders is a balancing game between

- Quality,
- bit rate,
- delay and complexity.

The quality is a function of the bit rate. For telephone quality speech the standard is 8 bits mulaw per sample. Using a 8 kHz sampling rate this results in 64 kb/s.

Speech coding algorithms can maintain this quality at substantially lower rates all the way down to 16 kb/s. At lower rates there will be some loss in quality, but even to rates as low as 1200 bits/s the speech is still quite intelligible

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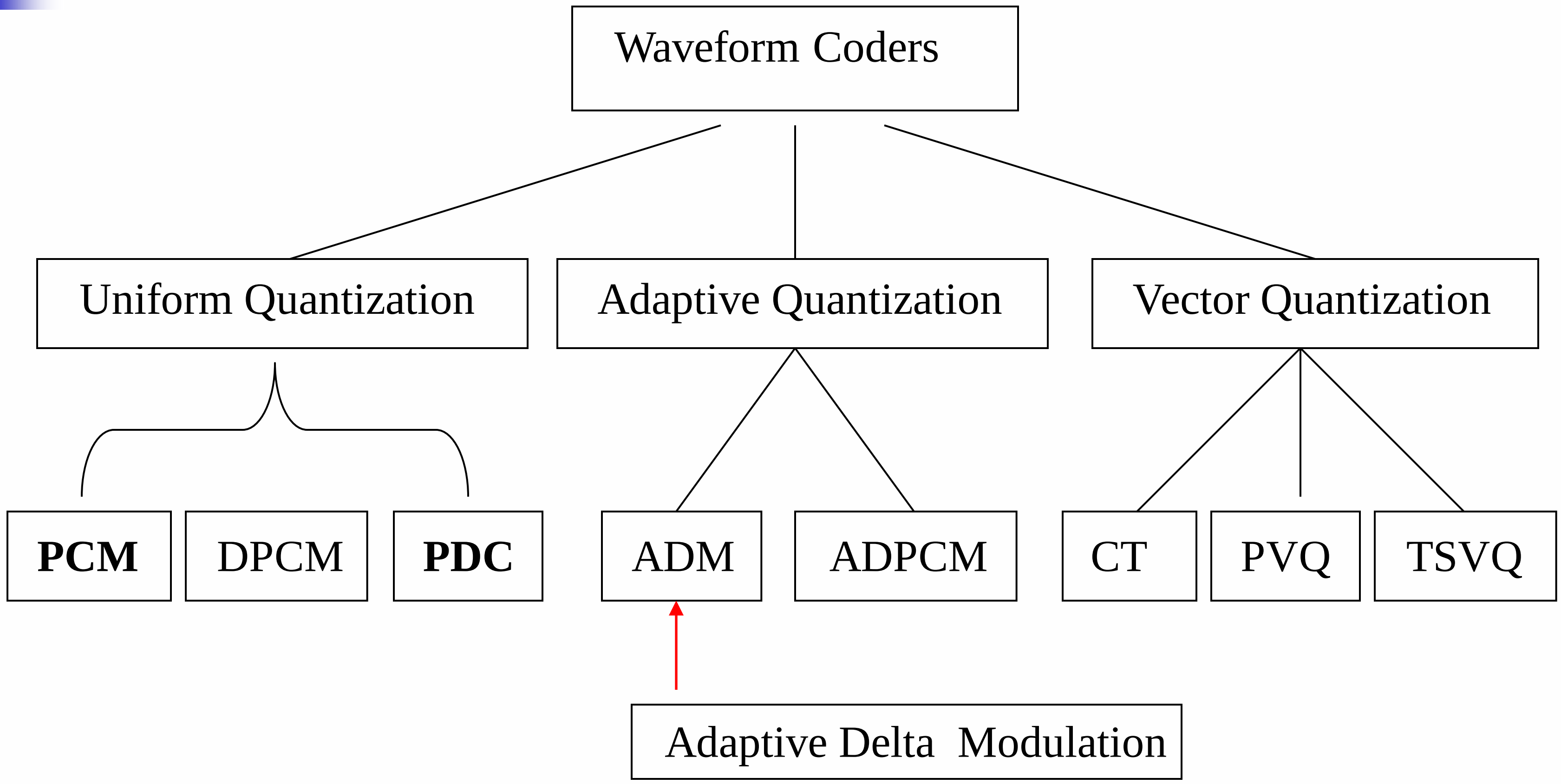
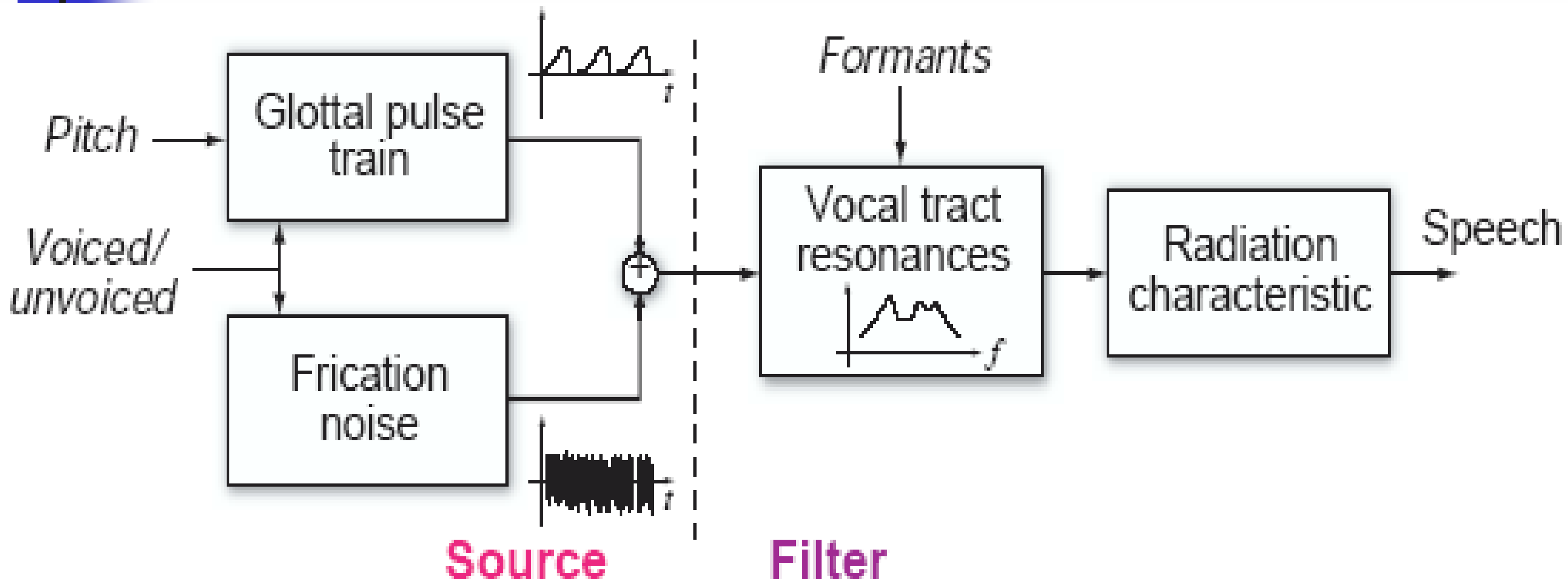
Introduction to Speech Coding

- Speech and audio compression has advanced rapidly in recent years spurred on by cost effective digital technology and diverse commercial application. It is including:
- 1. Waveform coding
- 2. Voice coding

Waveform Coders



Vocal tract and sound source modeling



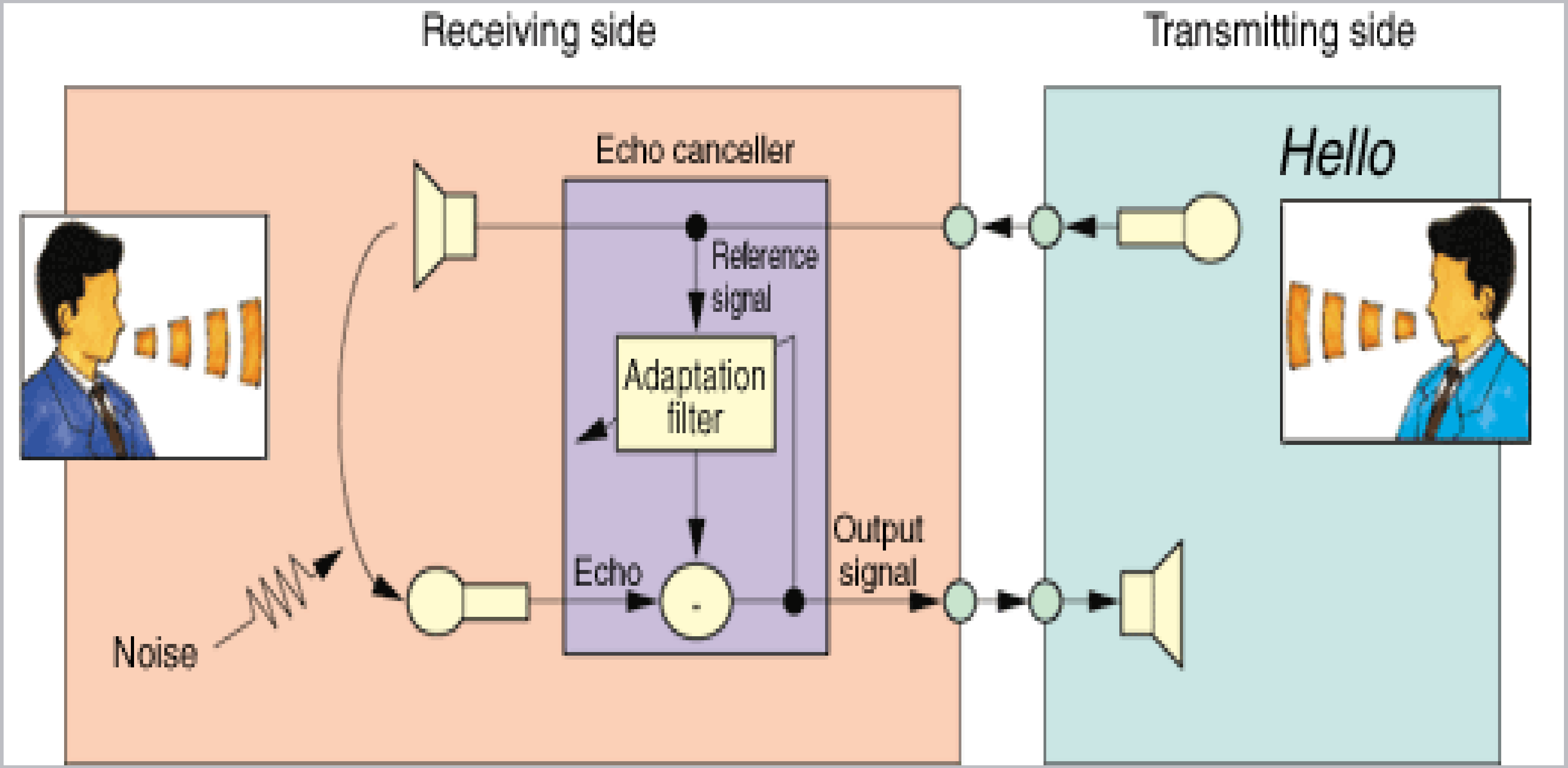
Acoustic

- *Acoustics* is the science of sound and the study of sound production and propagation.
- *Electro-acoustics* focuses on the transfer of a signal between acoustical and electrical form. It includes microphones and loudspeakers, echo cancellation, acoustic noise control, 3D audio and virtual acoustic audio rendering.

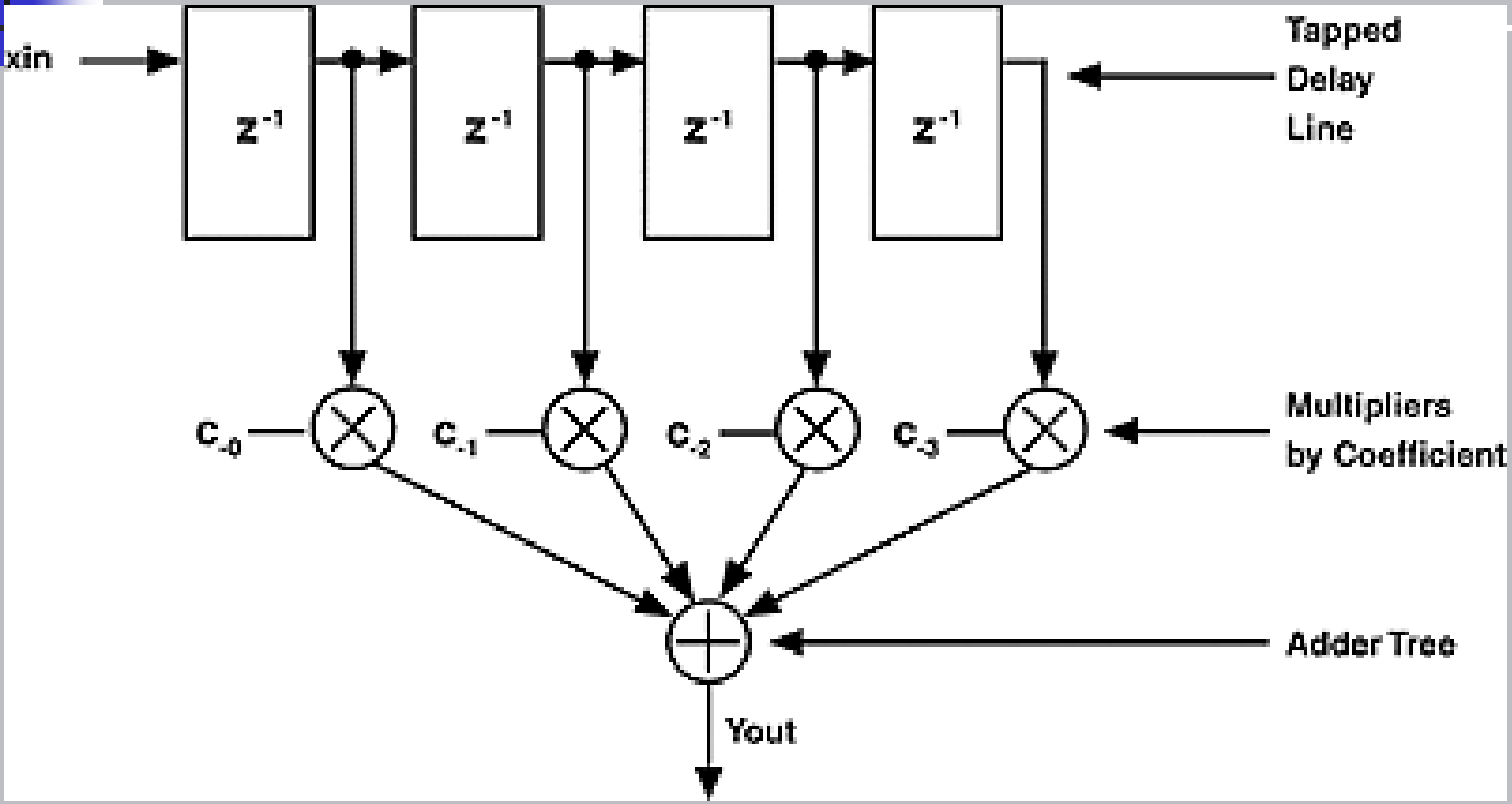
Echo Cancellation

- Echo cancellation was invented at Bell Labs in 1965 and research on network and acoustic echo cancellation continues. We have extended our investigations to the multi-channel problem and have successfully demonstrated real-time stereo acoustic echo cancellation in a teleconferencing system.

Echo Cancellation

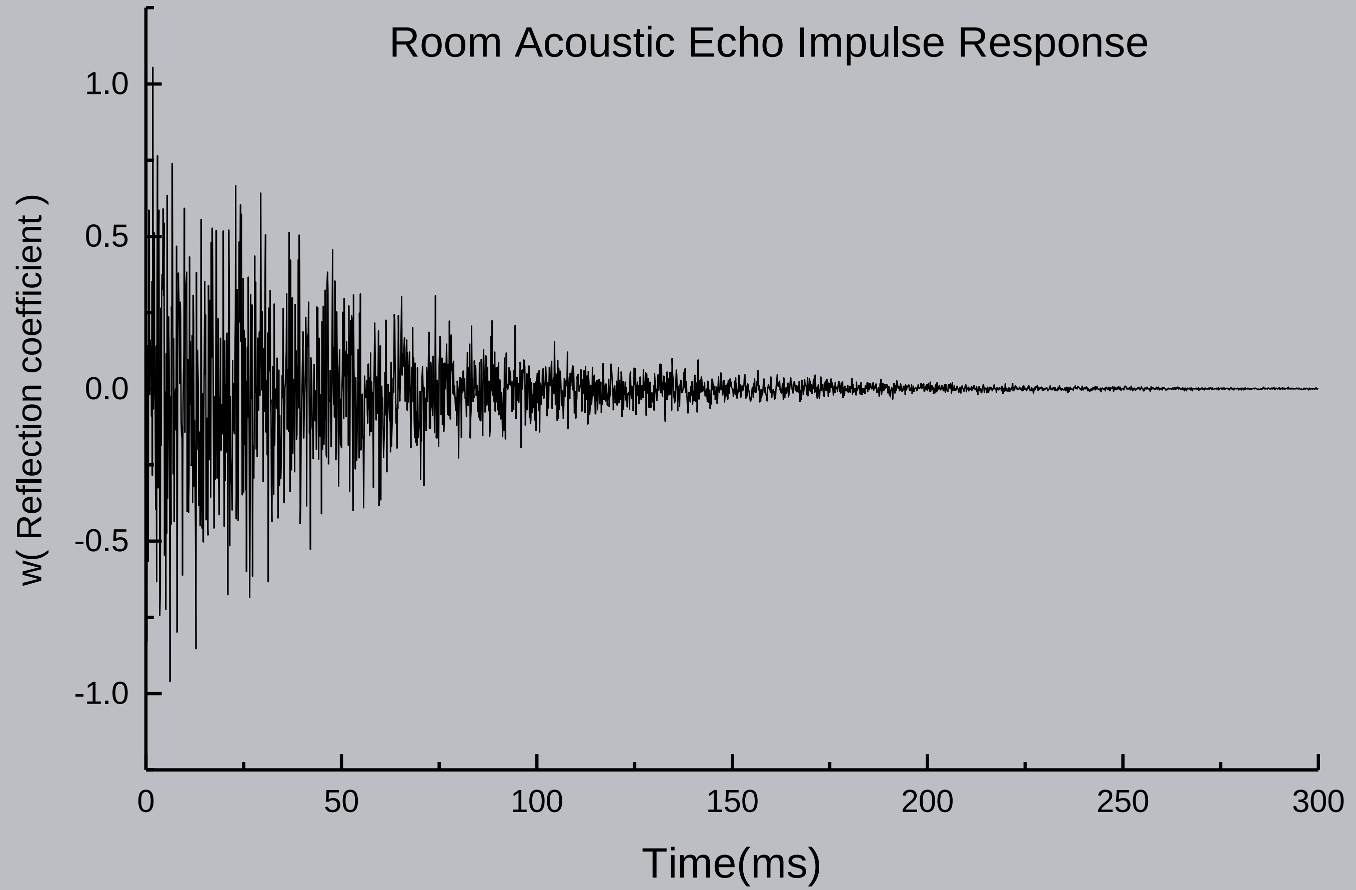


FIR Filter



- Basic FIR filter

Model for Acoustic Echo Impulse Response



Listening to Effect of Echo

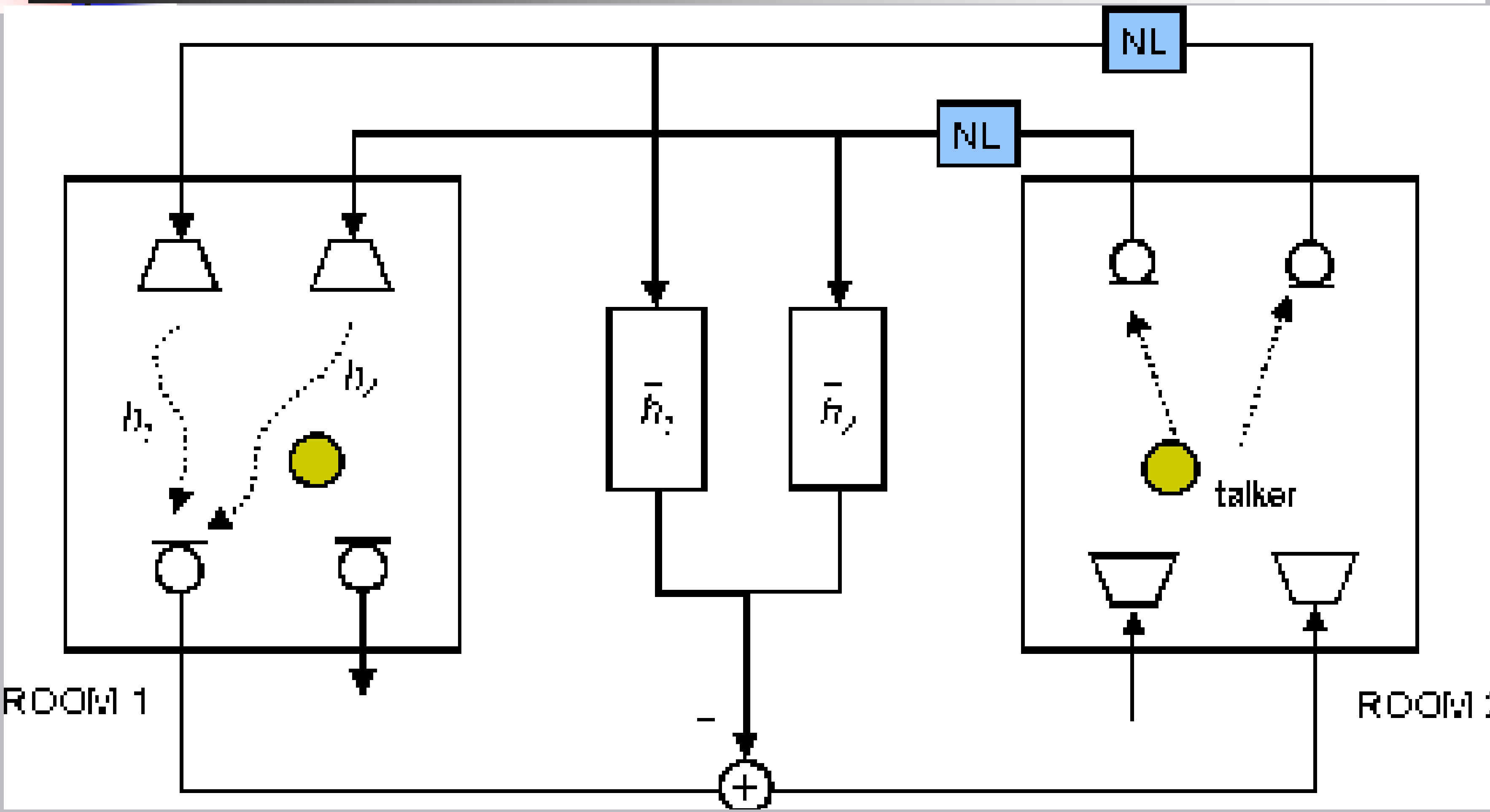
- Original Speech Signal

  Engilsh *Persian*

- Echo with 250 msec path

  Engilsh *Persian*

Structure of the stereophonic echo cancellation



Active Acoustic Noise Control

- Active noise control generates an opposing wave that is equal in amplitude but out of phase with the acoustic noise to be reduced.
- Active noise control makes use of adaptive digital filters in conjunction with reference and error sensing transducers and a secondary source, usually a loudspeaker.

This technology has been used in the control of noise generated in heating, ventilation, and air conditioning (HVAC) ducts, automobile exhaust noise, and aircraft engine and propeller noise, to mention just a few applications.

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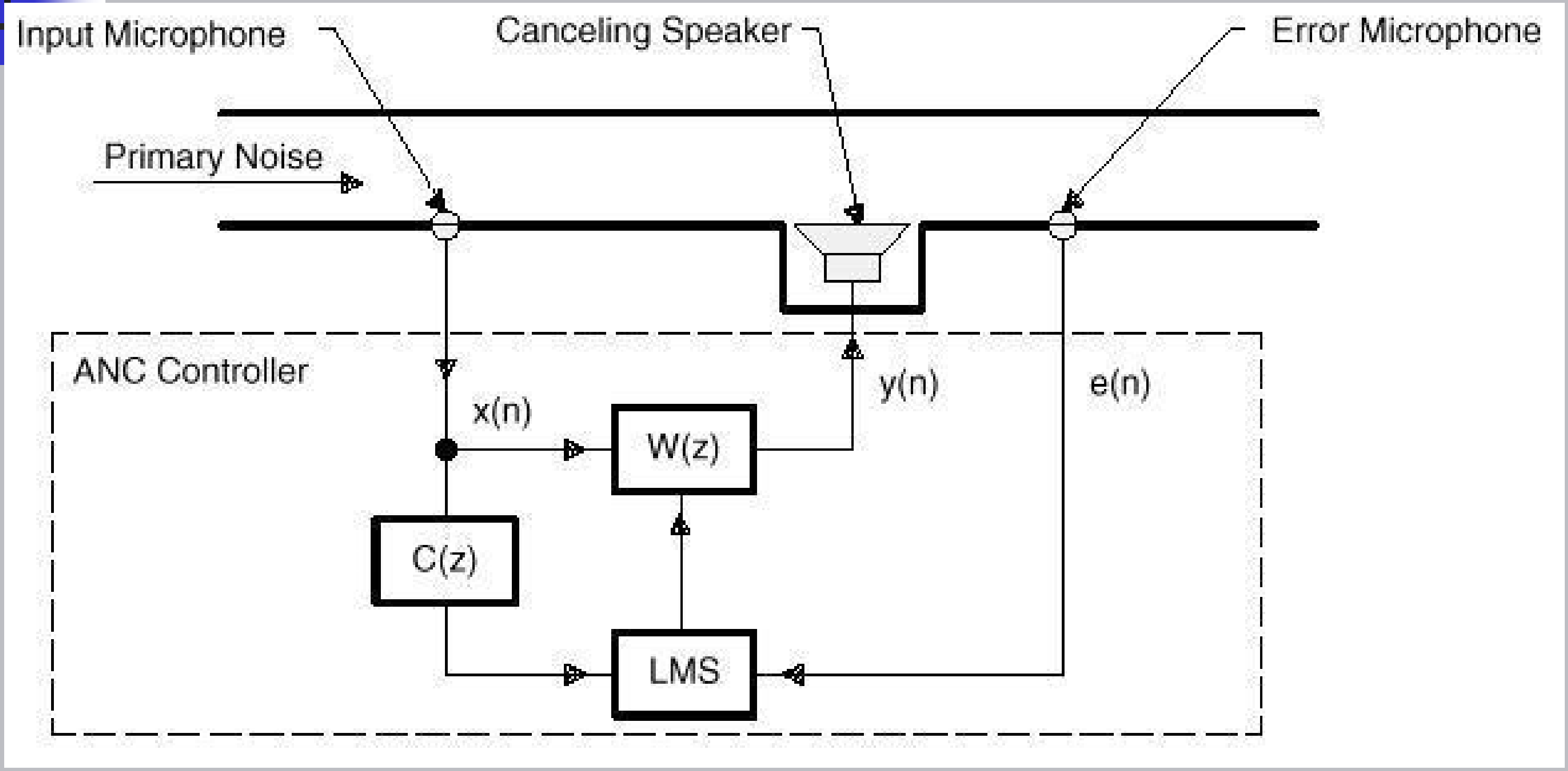
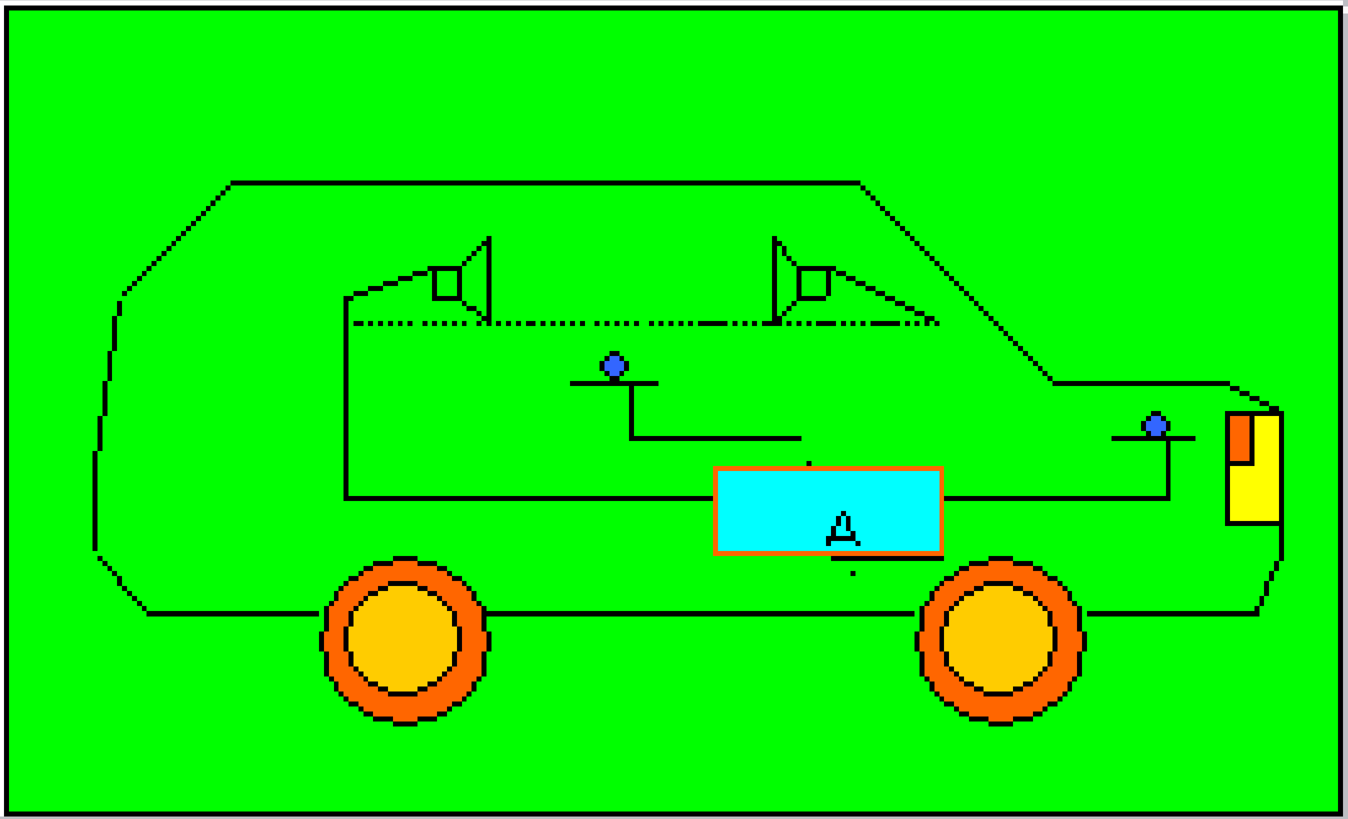
FXLMS Algorithm



- Acoustic noise control using the FXLMS algorithm

Acoustic Noise Control

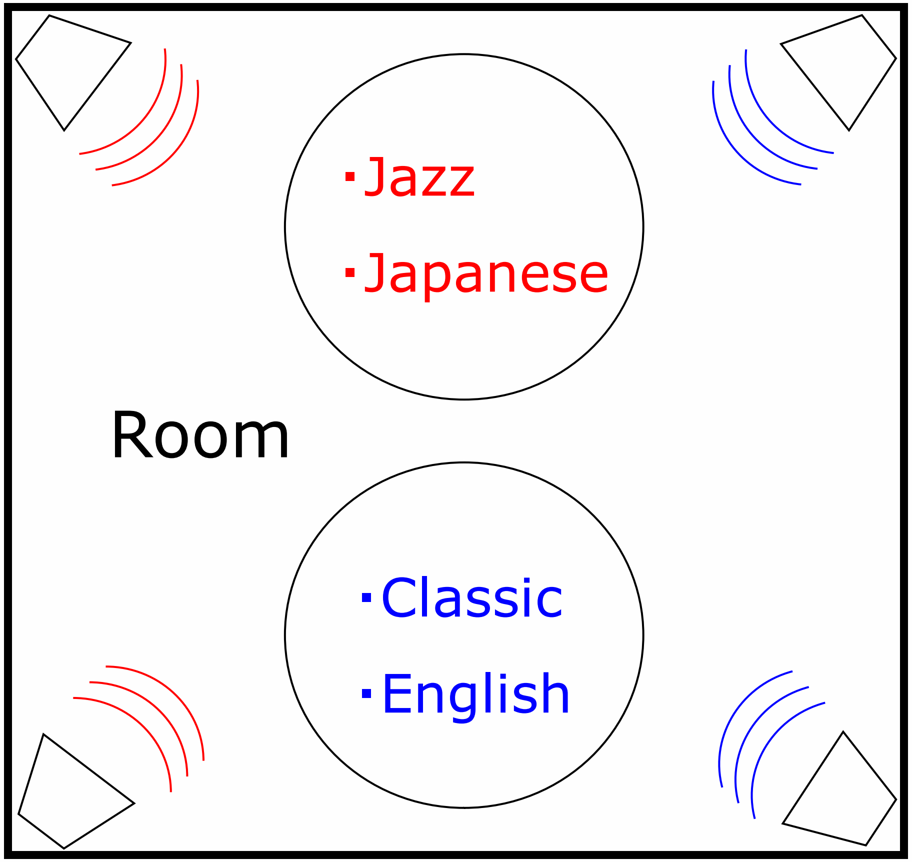


- Acoustic noise control in cars

Smart Acoustic Room (SAR)

- Smart Acoustic Room (SAR) is defined the acoustic response between two (or more) points could be controlled smartly. By control, we mean to have a well estimation of the acoustic path between two points and then to make the appropriate signal to cancel an unwanted noise or to emphasis to a desired signal (speech or music).

Application of SAR



When there are the peoples who want to listen to Jazz or Classic in a room, we don't want to use headphone as it totally isolate the person from surrounding.

In a conference room or big hall, we have two kinds of audiences that want to listen to the Japanese or English speech. If we can give two audiences the desire location, just by seating in the right place one can hear to desire language.

Application of SAR

- We proposed a new type echo canceling by using the SAR system. The new algorithm uses two speakers and one microphone, by smartly control the acoustic impulse response the speaker signal will be cancelled at the microphone position locally. That is, the microphone cannot receive any echo signal. .

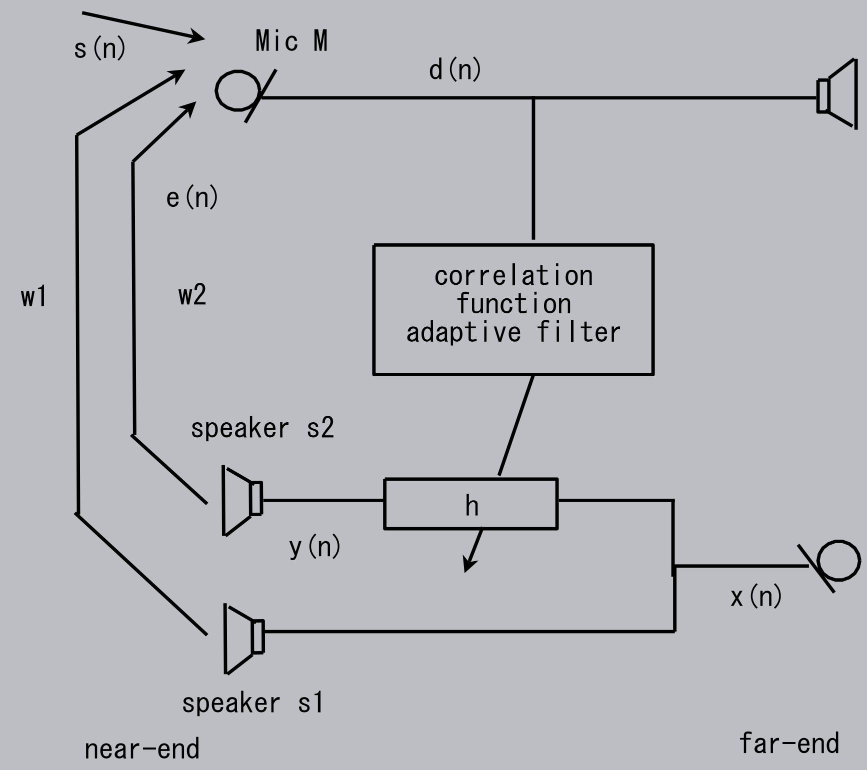


Fig.2 Echo canceling by using the SAR system

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Experiment results

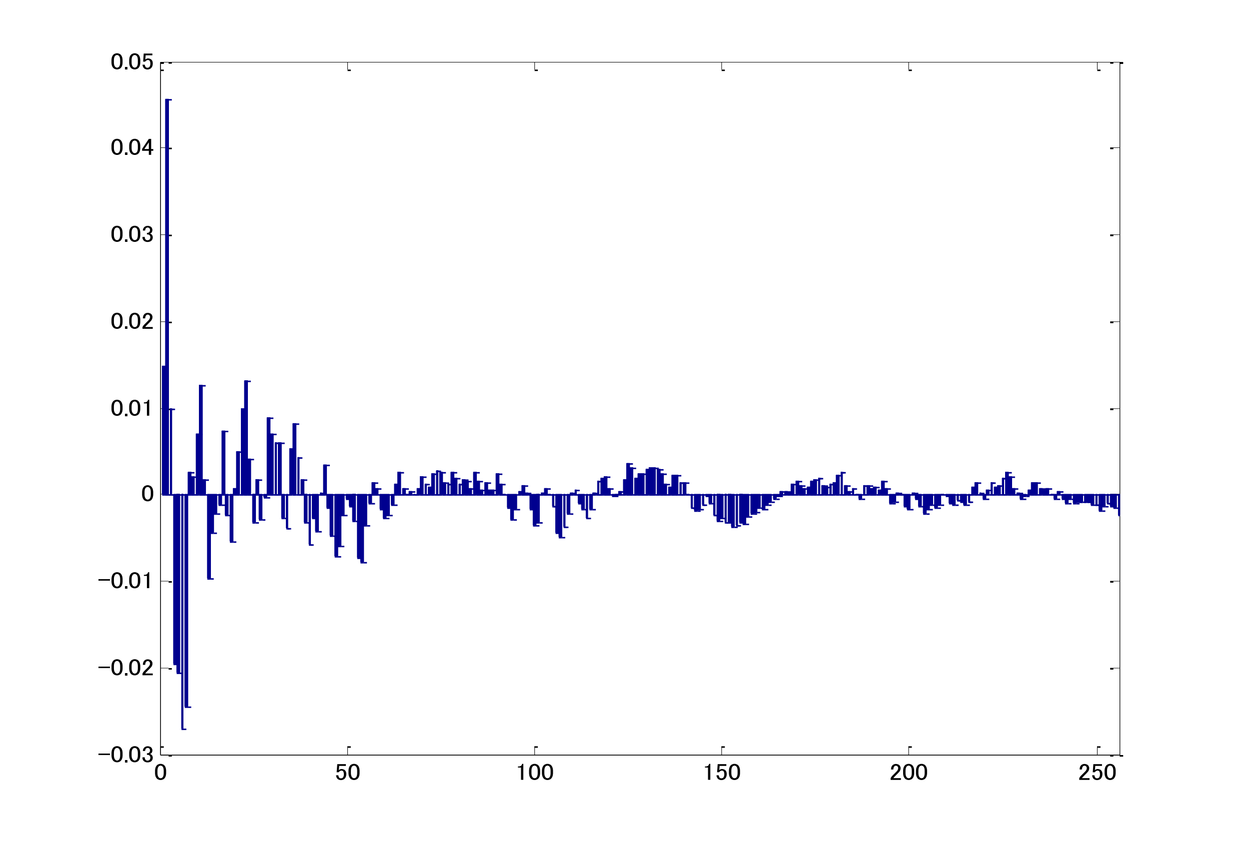


Fig.7 Example of the measured impulse response (from speaker S1 to 5-5 position)

Experiment results

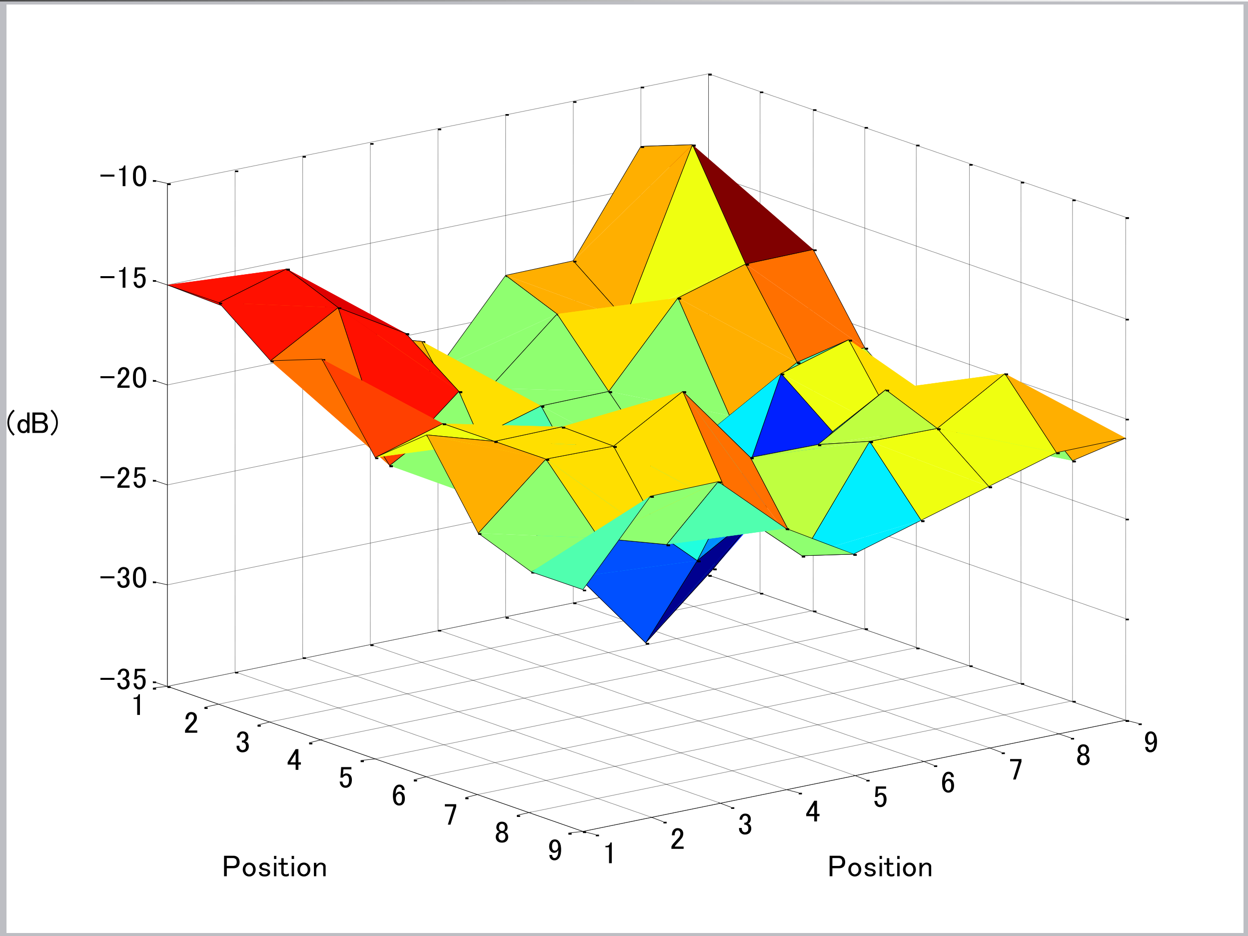
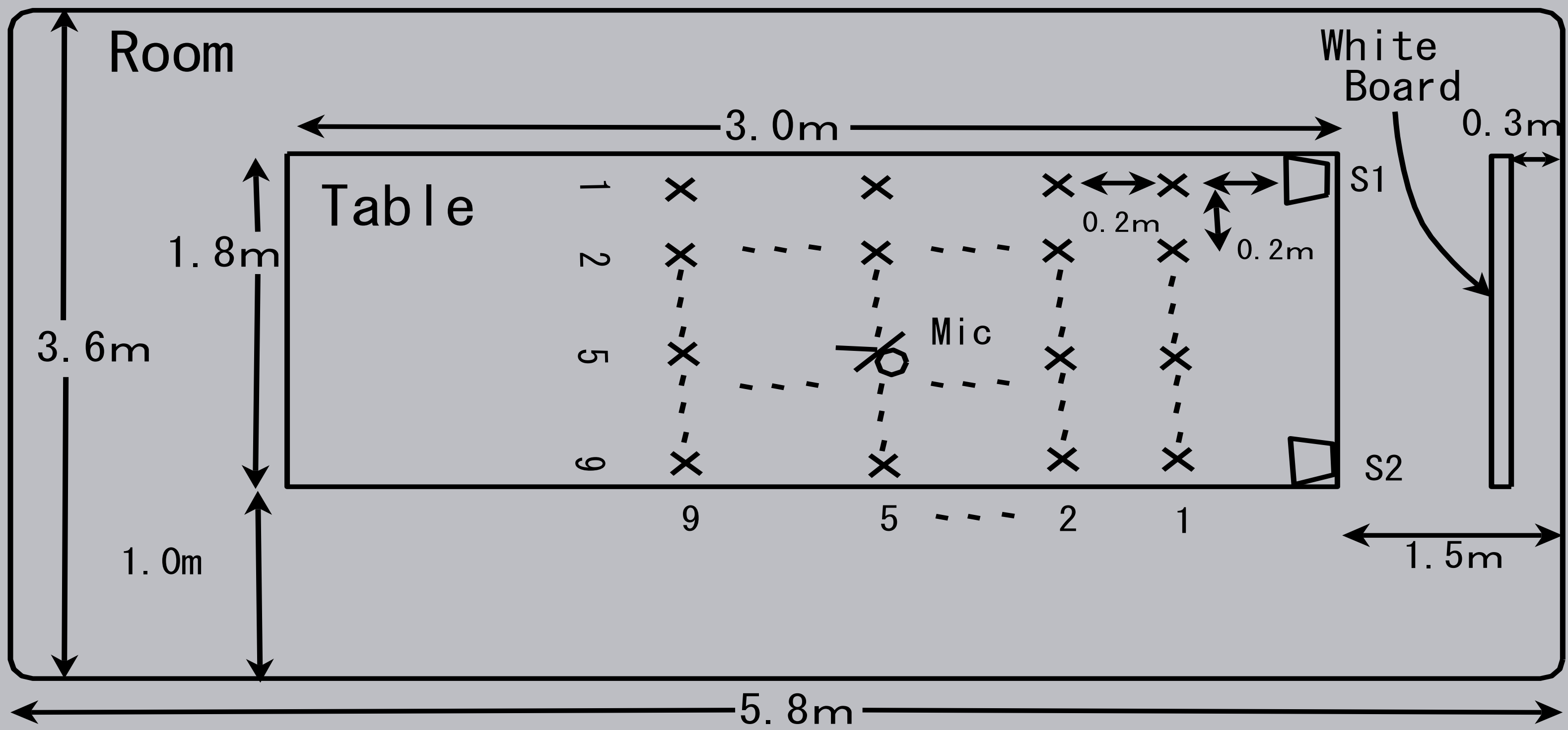


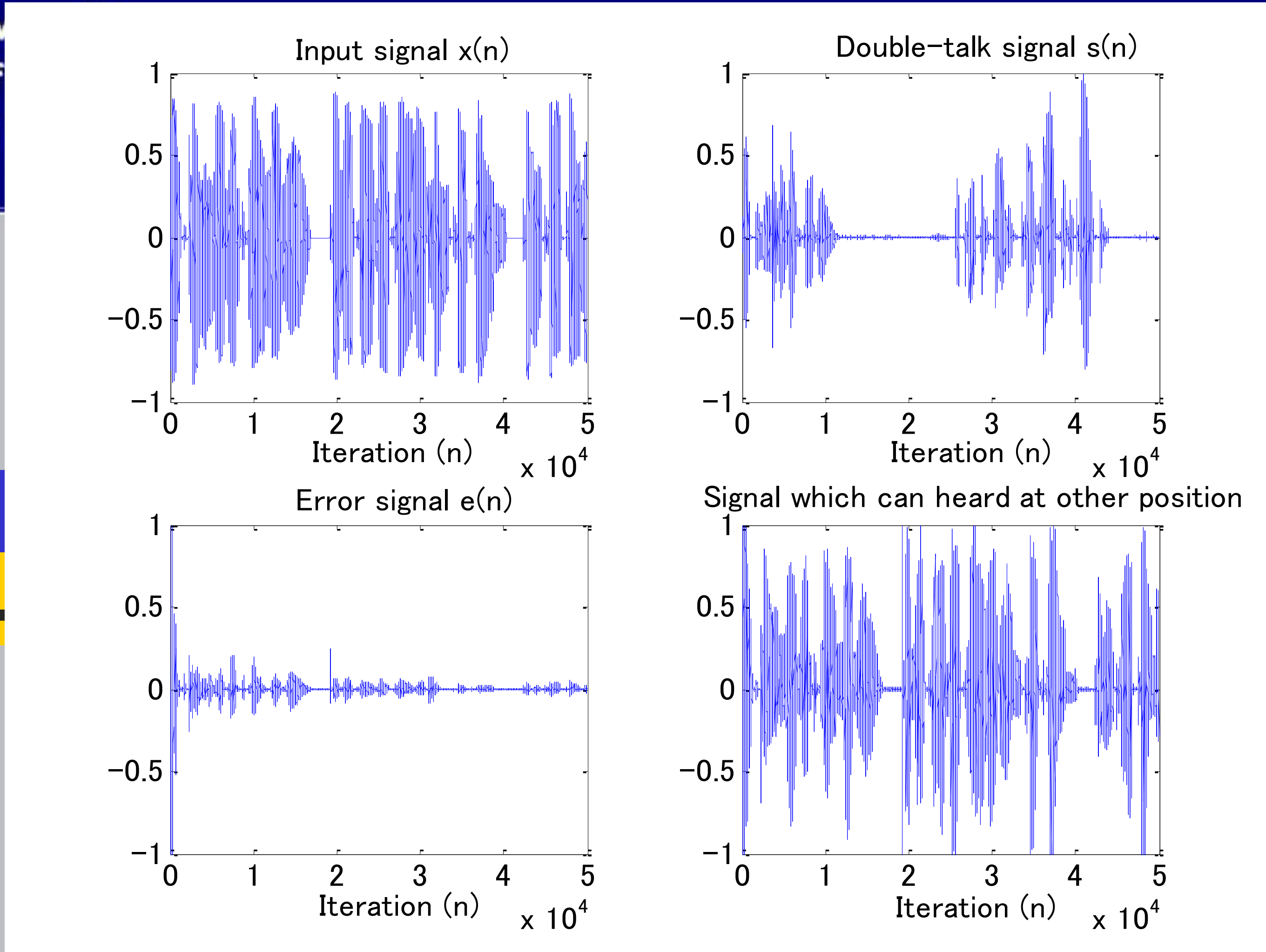
Fig.8 the 3D hole of the MSE

Experiment results



RoomSize 5. 8m × 3. 6m × 3. 0m

Fig.6 Experiment environment



Input signal (x(n))

Double-talk signal (s(n))

Signal at the Mic position (d(n))

Signal at the other position

Back

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Simulation results

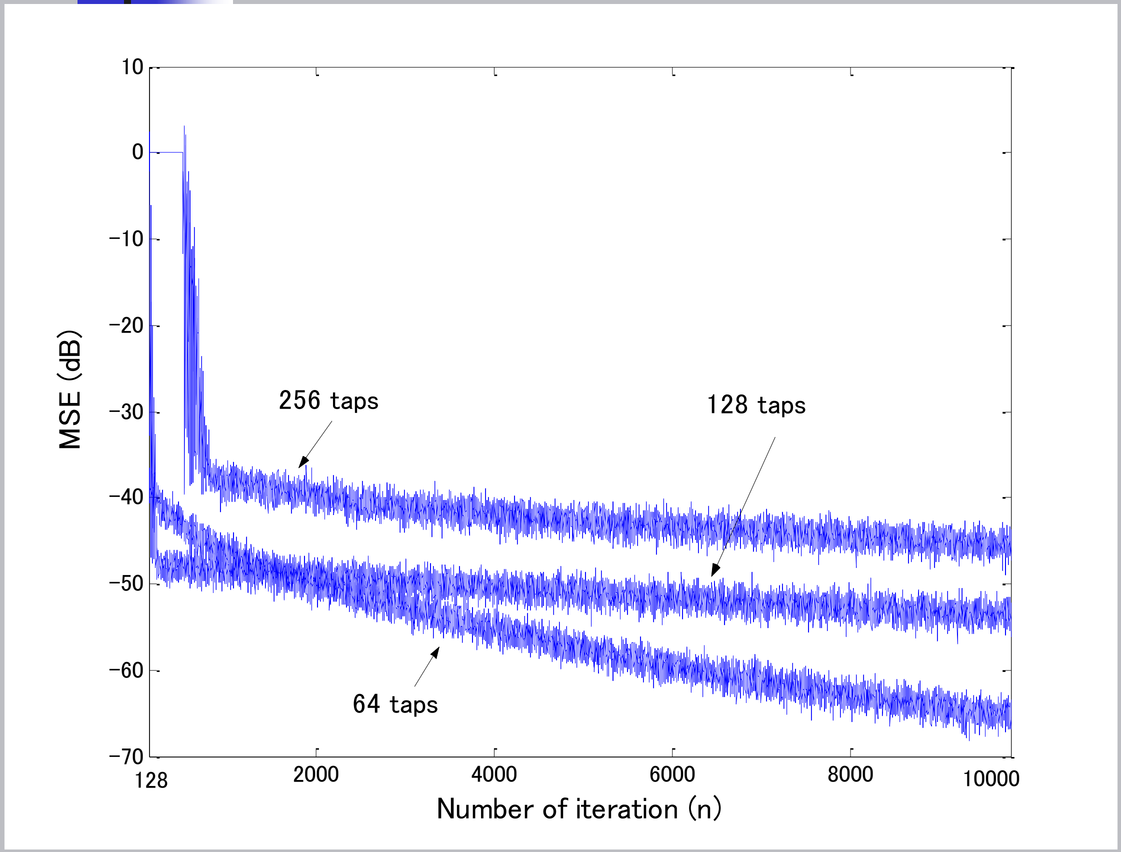


Fig.5 The MSE of the SAR algorithms by using virtual Mic

- The input signal *x(n)* is the white noise. Each computer simulation is executed 100 times.

SAR algorithm by using virtual microphone

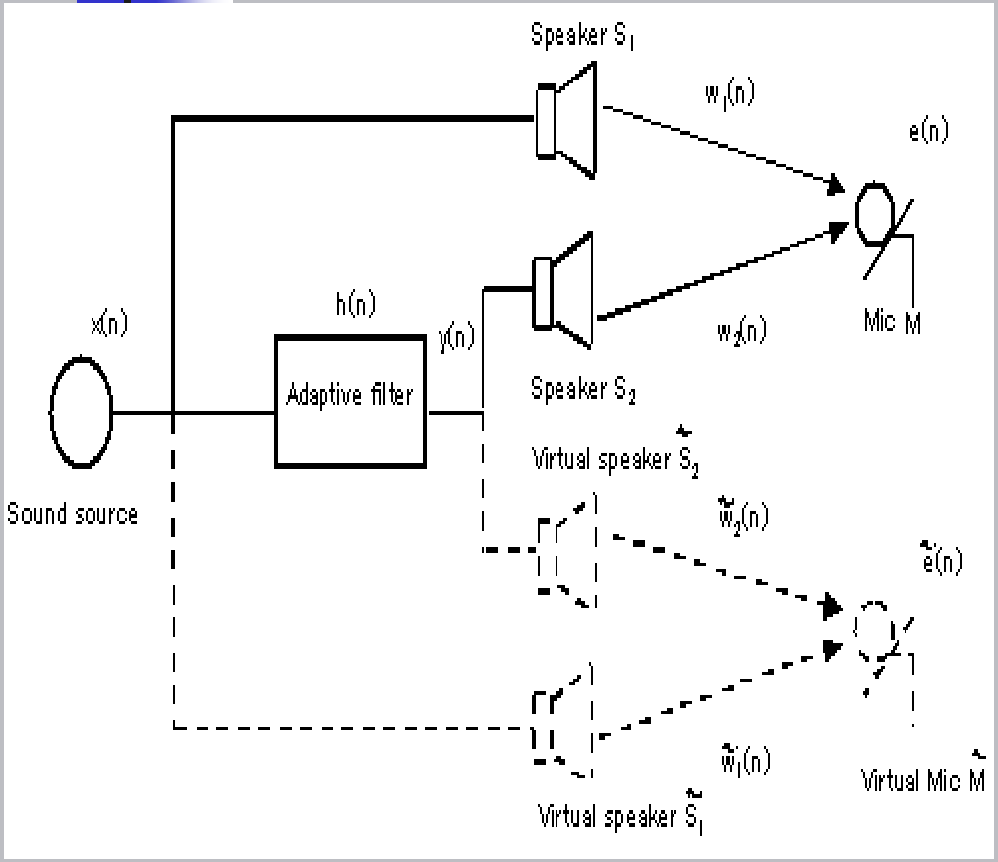


Fig.4 SAR model by using the virtual microphone

- Error Signal:

$$e(n) = x(n) * w_1(n) + x(n) * h(n) * w_2(n) \quad (1)$$

- Virtual error Signal:

$$\tilde{e}(n) = x(n) * \tilde{w}_1(n) + x(n) * h(n) * \tilde{w}_2(n) \quad (2)$$

Two-speakers SAR system

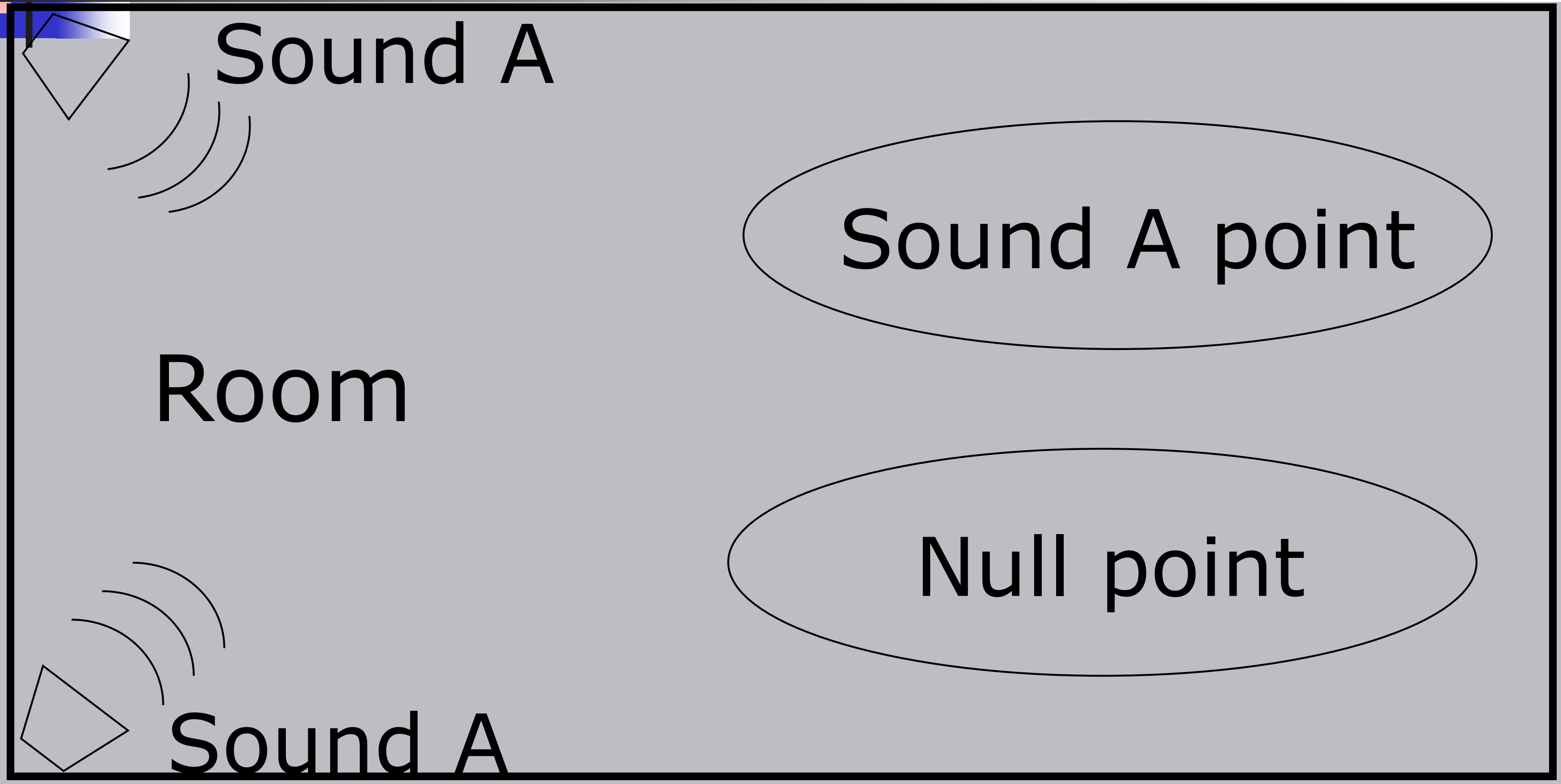
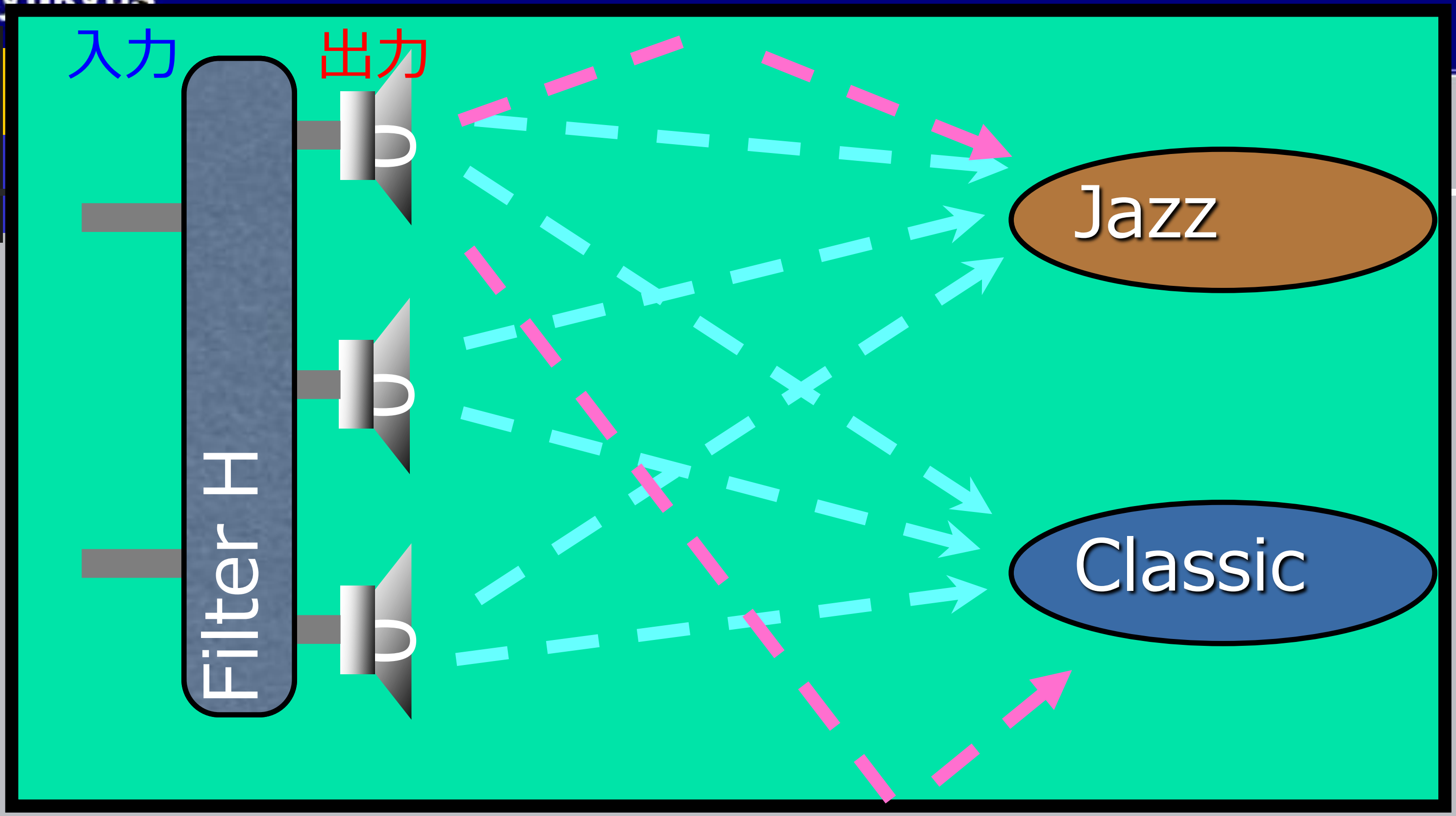


Fig.3 Two-speakers SAR system



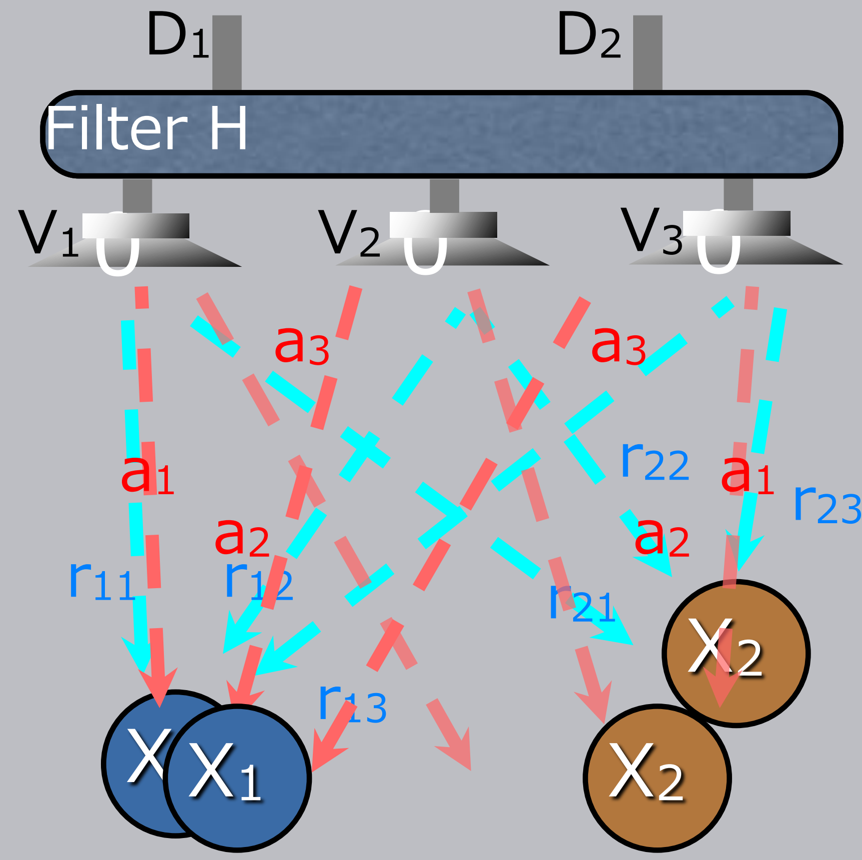
Acoustic path

Adaptation for one's location to listen to the desired sound.

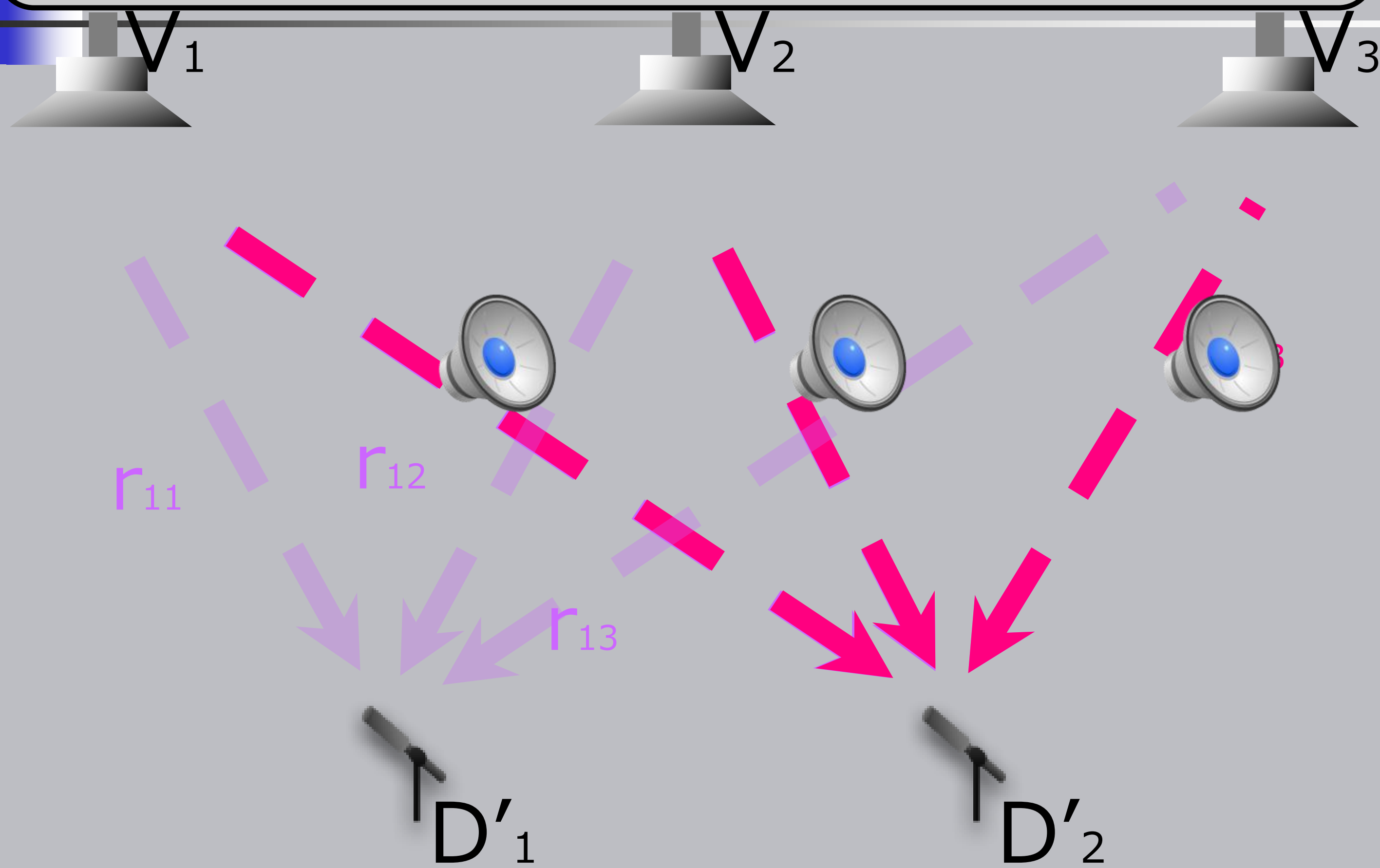
3 Speakers System

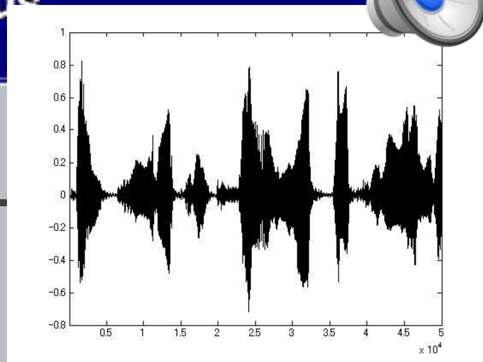
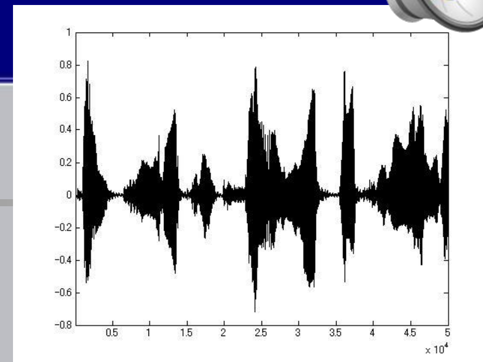
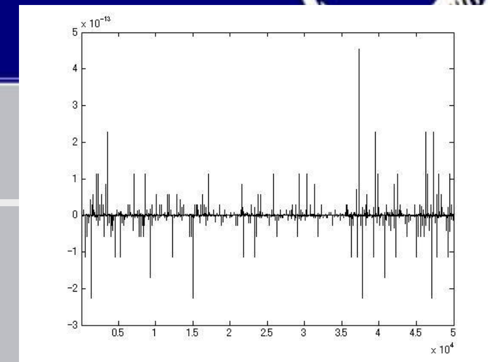
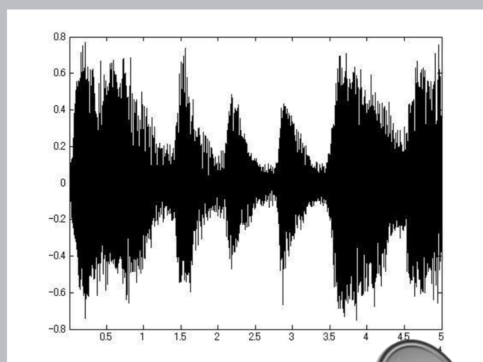
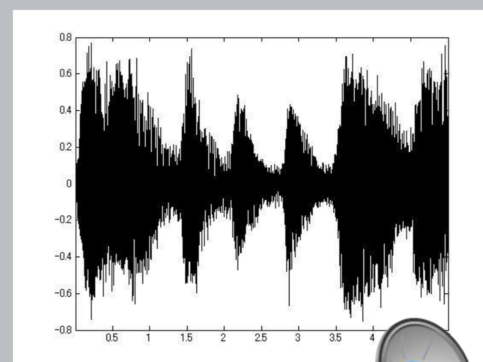
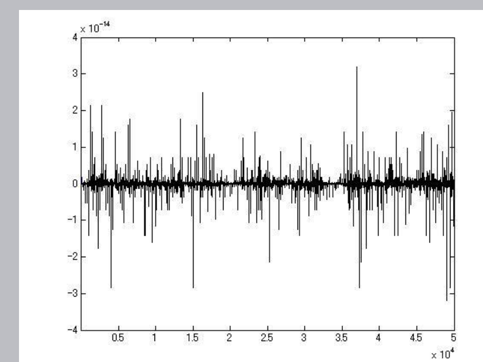
$$A = \begin{bmatrix} r_{11} & r_{12} & r_{13} \\ r_{21} & r_{22} & r_{23} \end{bmatrix}$$

$$= \begin{bmatrix} a_1 & a_2 & a_3 \\ a_3 & a_2 & a_1 \end{bmatrix}$$



Crosstalk Cancellation Filter H



	D	D′	$e = D' - AHD$
Signal 1			
Signal 2			

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Blind Source Separation (BSS) Study by ICA Based on Information Maximization Method in the CPP & Reflected-Overlapped Images

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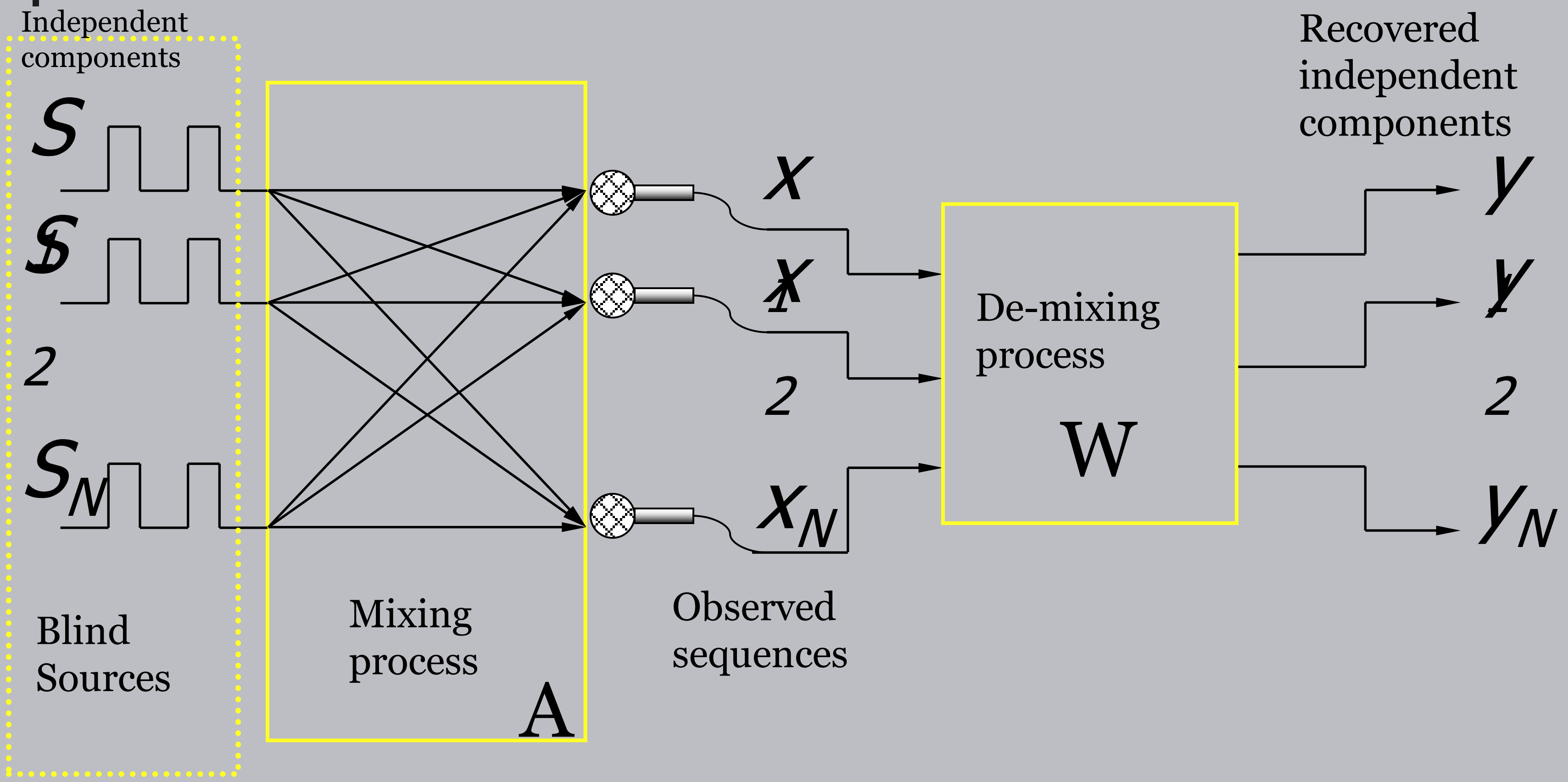
Assume observation signals obtained by N sensors are linear mixing of N unknown independent *source* signals!



How BSS works?

APPLICATIONS of BSS

Noise Elimination in general

Speech Processing (Cocktail Party Problem, Noisy environment,...)

Sonar, Radar

Sismic waves

Preprocessing recognition

Image Processing

Reflected-Overlapped Images

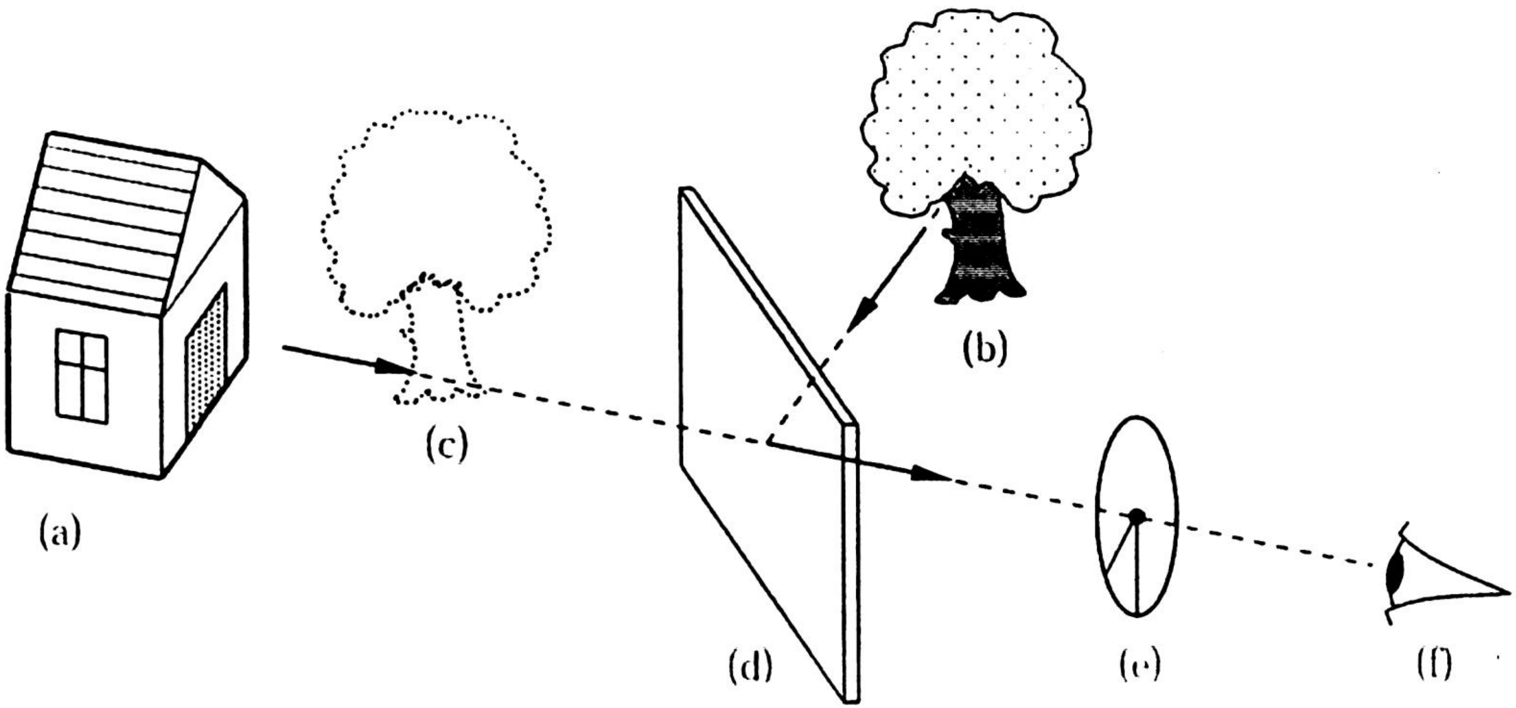


Figure 1 – A typical optical setup including a semireflector: (a) – object 1, (b) – object 2, (c) – virtual object, (d) – glass, (e) – polarizer, (f) – camera.

The mixture matrix

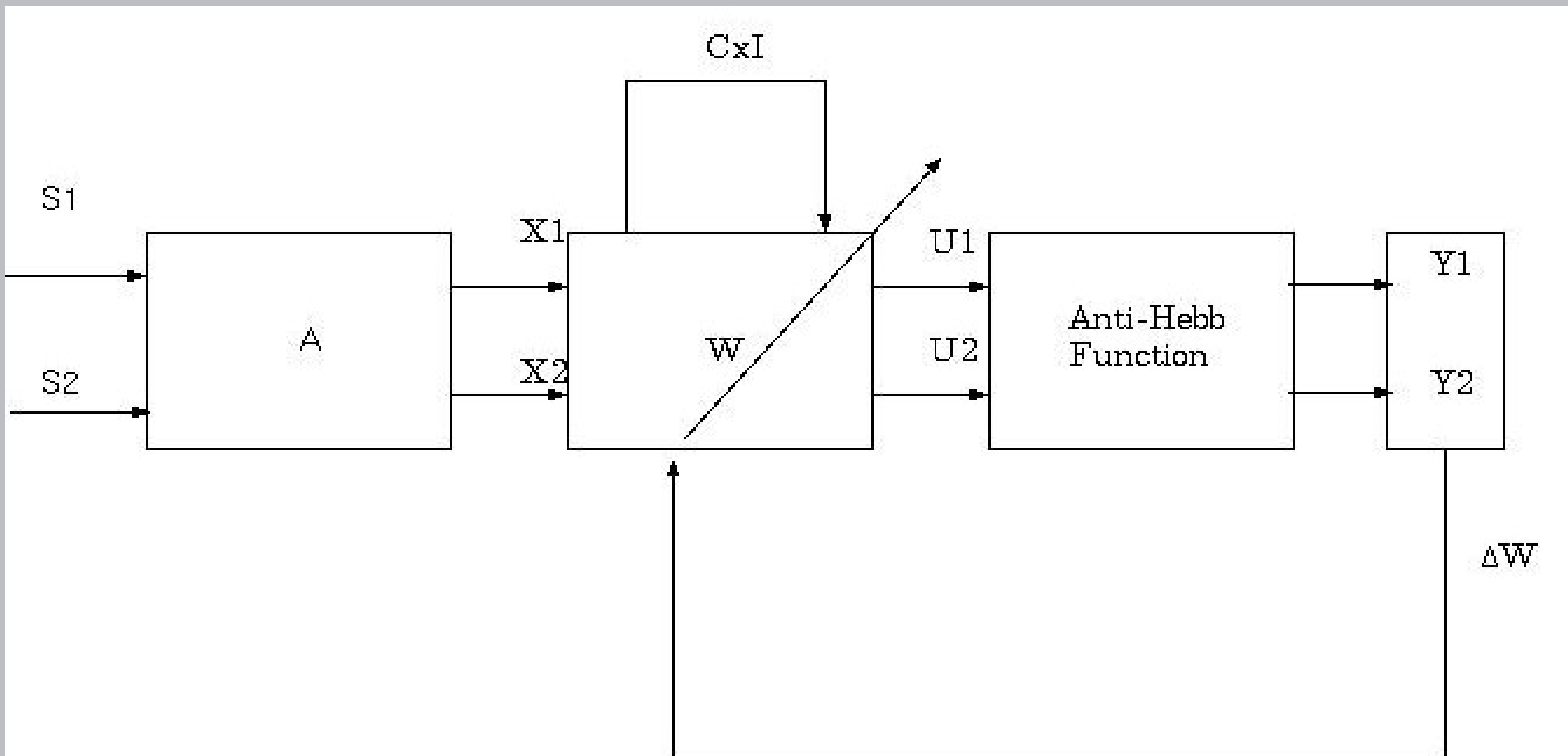
$$x_1 = a_{11}s_1 + a_{12}s_2$$

$$x_2 = a_{21}s_1 + a_{22}s_2$$

$$X = AS$$

$$A = \begin{bmatrix} a_{11} & a_{12} \\ a_{21} & a_{22} \end{bmatrix}$$

The algorithm of simulation



The algorithm of simulation

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Independent Component Analysis

The statistical independence of separated sources can be measured using the Kullback-Leibler (KL) divergence between the product of the joint density and marginal densities:

$$\int P_{\mathbf{Y}}(\mathbf{Y}) \log\left(\frac{P_{\mathbf{Y}}(\mathbf{Y})}{\prod_{i=1}^{n} P_{Y_i}(Y_i)}\right) d\mathbf{Y}$$

and the independence is achieved if and only if this KL divergence is equal to zero, i.e.

$$P_{\mathbf{Y}}(\mathbf{Y}) = \prod_{i=1}^{n} P_{Y_i}(Y_i)$$

It is equivalent to minimize the mutual information

$$I(\mathbf{Y}) = \sum_{i=1}^{n} H(Y_i) - H(\mathbf{Y})$$

The best proposed learning rule for the above goal is Natural Gradient Learning Algorithm (NGLA). NGLA has an iterative hardware friendly structure.

$$\boldsymbol{W}_{t+1} = \boldsymbol{W}_t + \mu[\boldsymbol{I} - \boldsymbol{g}(\mathbf{Y})\mathbf{Y}^T]\boldsymbol{W}_t$$

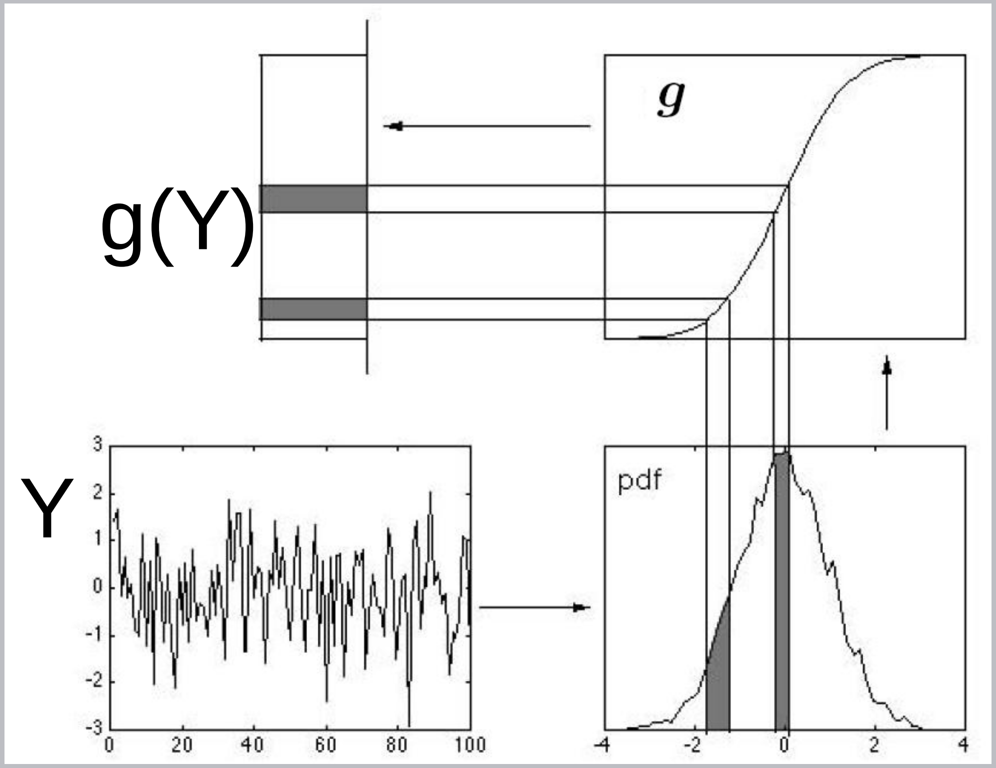
g(.) is activation function.

Kullback Leibler ICA

$$W_{t+1} = W_t + \mu[I - g(Y)Y^T]W_t$$

A brief explanation about the above learning equation:

g is sigmuid function which transform a signal set with gaussian distribution to signals set of monotonic distribution. In other words: if Y is a a set of signals with gaussian PDF, then g(Y) will be a set of signals with monotonic PDF.



Stone BSS

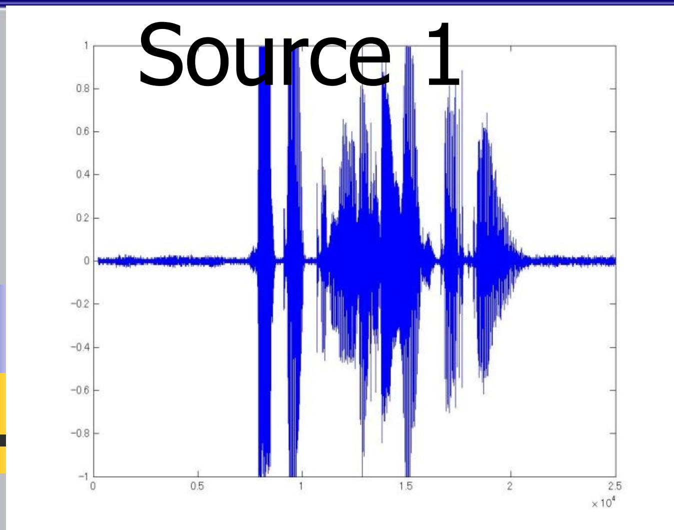
- Stone BSS conjecture indicates that the temporal predictability of any signal mixture is less than (or equal to) that of any of its component source signals. Stone's measure of temporal predictability for a N-sampled signal has been defined as follows

$$F_{classic}(y) = log\frac{V_i}{U_i} = log\frac{\sum_{n=1}^{N}(y_{long}(n) - y(n))^2}{\sum_{n=1}^{N}(y_{short}(n) - y(n))^2}$$

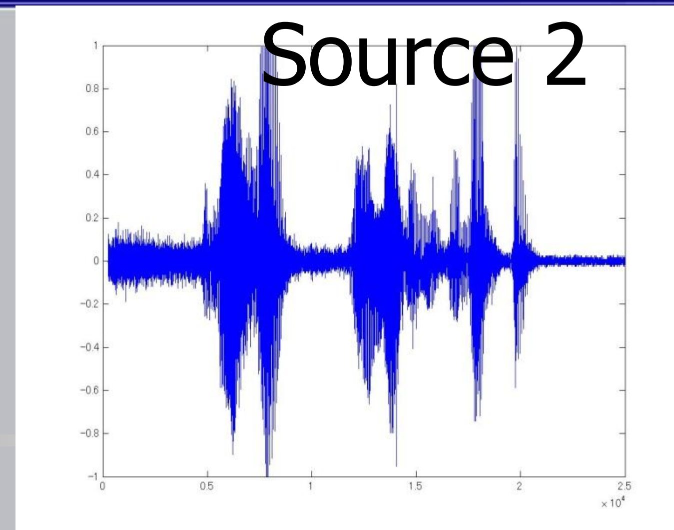
- Stone's BSS is transferred to generalized eigenvalue decomposition. It has the complexity order of $O(M_R^3)$

Simulation result of speech

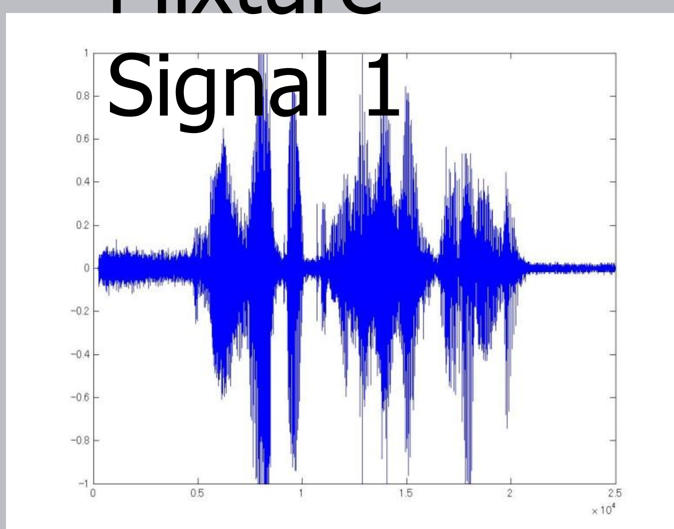
Source 1



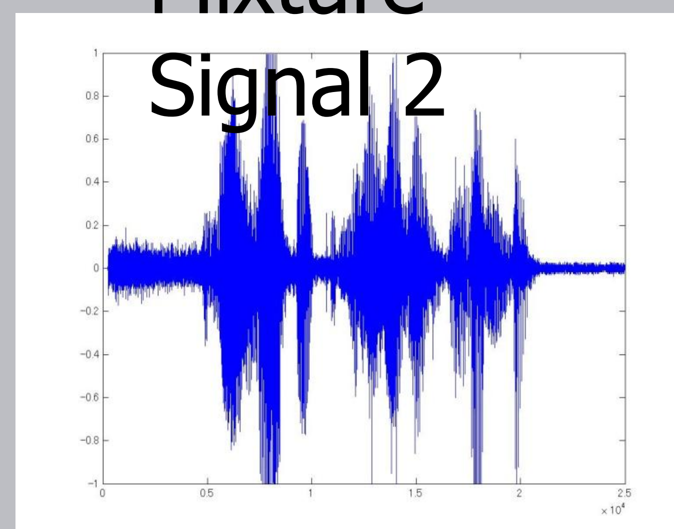
Source 2



Mixture Signal 1

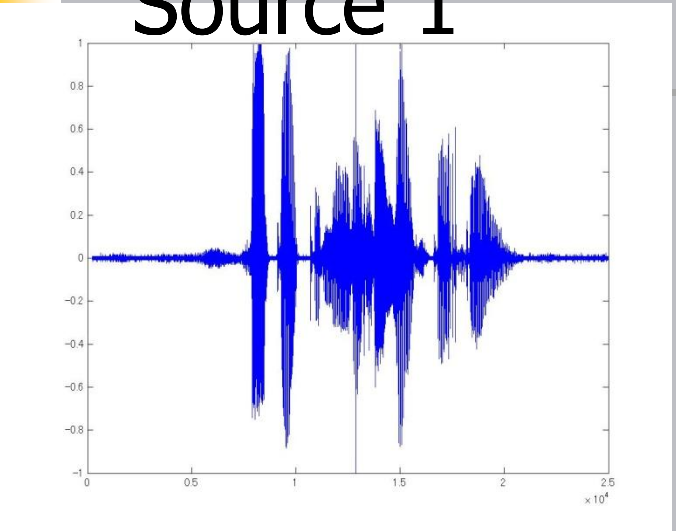


Mixture Signal 2

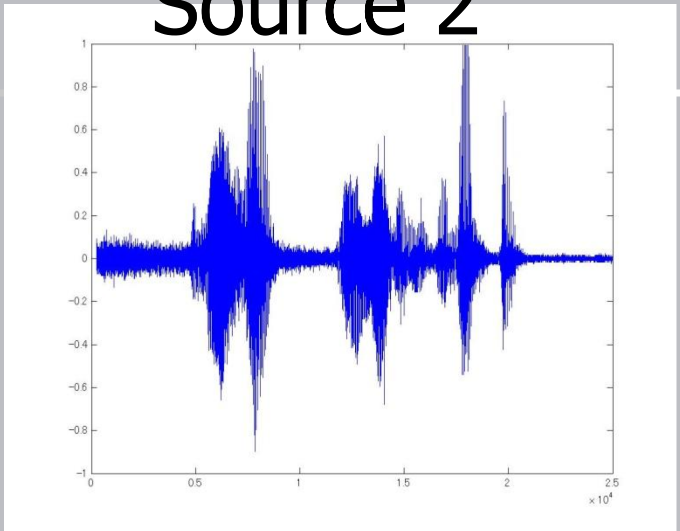


Simulation result of speech

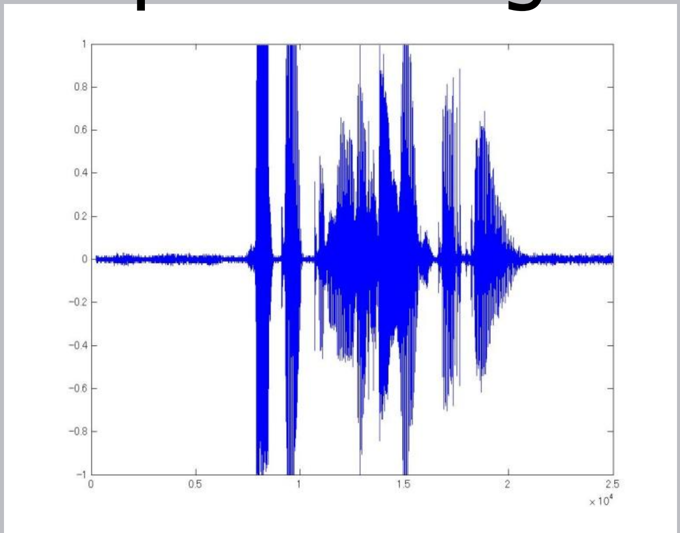
Source 1



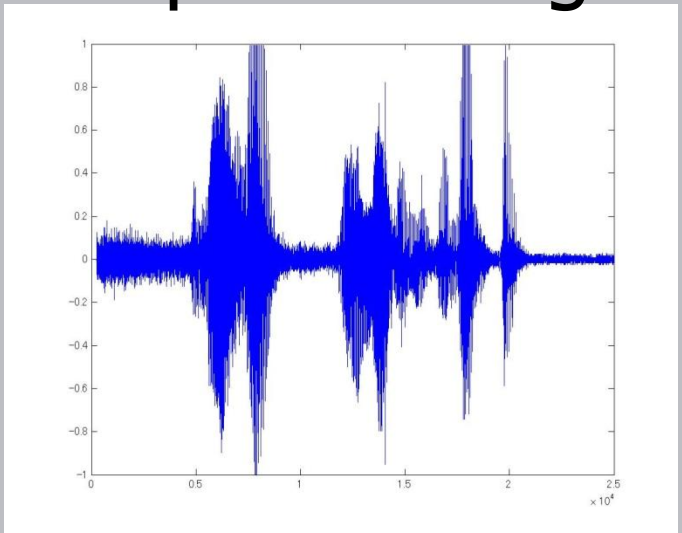
Source 2



Separated signal 1



Separated Signal 2



Real Speech Sounds

Source Speech 1

Source Speech 2

Mixed Speech 1

Mixed Speech 2

Separated Speech 1

Separated Speech 2

Simulation result of image

The Original Images

Simulation result of image

The Mixed Images

Simulation result of image

The Separated Images

Back

Digital Communications

- Digital communication techniques deal with transporting digital information (e.g quantized or/and compressed speech, audio, image and video) reliably from a source to a destination.

Digital Communications

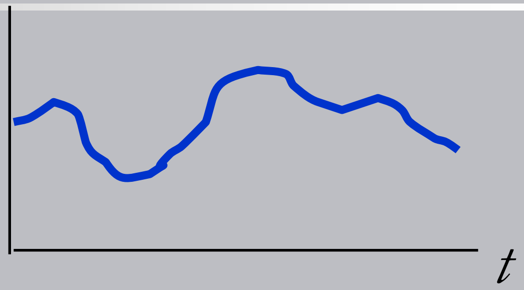
- Digital audio broadcasting
- Digital cellular communication
- Storage in multilevel memory cells
- Efficient algorithm mapping and implementation

Can we classify signals?

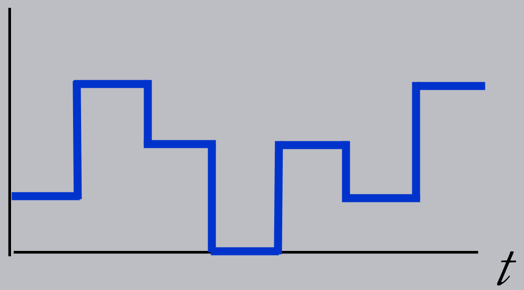
- Messages or signals can be classified:
- Analog
  - A physical quantity that varies with "time", usually in a smooth or continuous fashion
  - Fidelity describes how close is the received signal to the original signal. Fidelity defines acceptability
- Digital
  - An ordered sequence of symbols selected from a finite set of discrete elements
  - When digital signals are sent through a communication system, degree of accuracy within a given time defines the acceptability

Examples

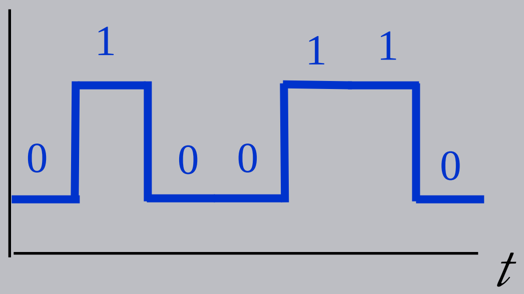
- Analog Signals
  - Values are taken from an infinite set



- Digital Signals
  - Values are taken from a discrete set



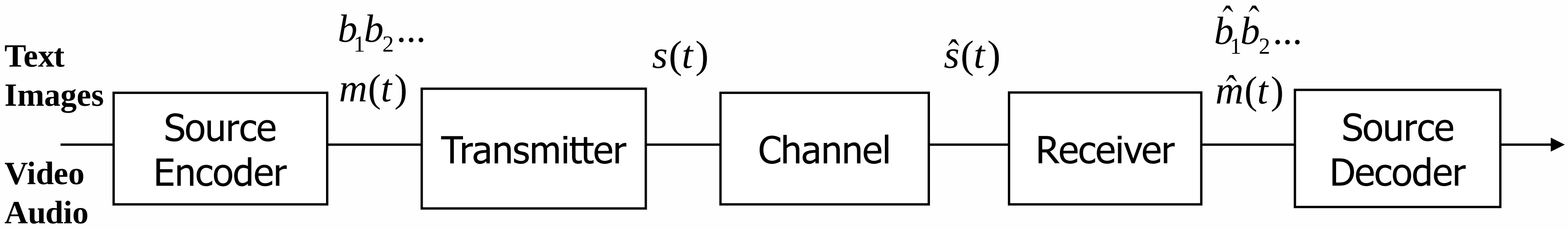
- Binary Signals
  - Digital signals with just two discrete values



Elements of Communication Systems

- Transmitter
  - Modulation
  - Coding
- Channel
  - Attenuation
  - Noise
  - Distortion
  - Interference
- Receiver
  - Detection (Demodulation+Decoding)
  - Filtering (Equalization)

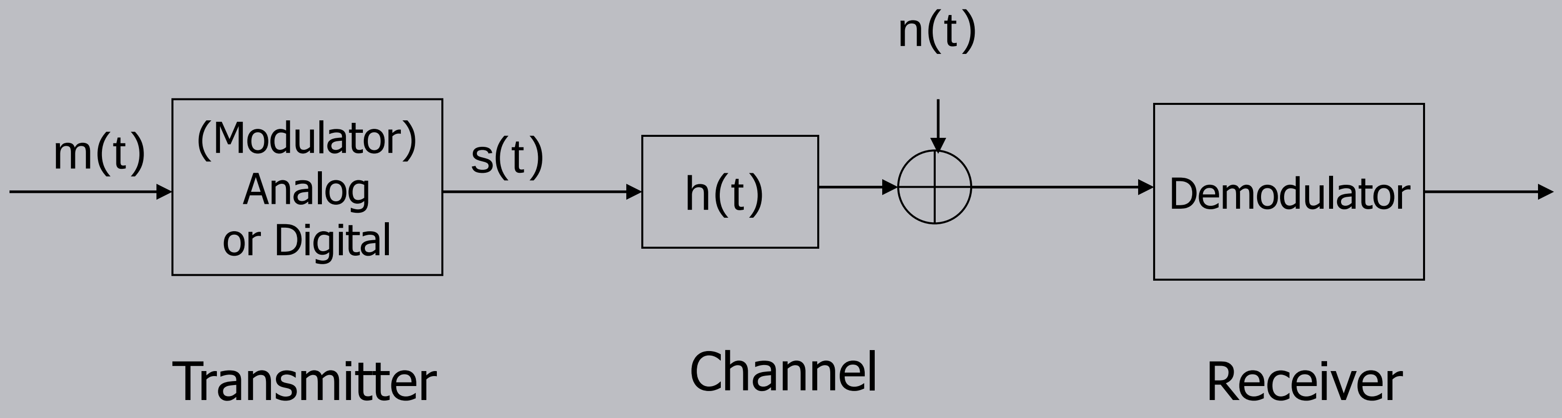
Elements of Communication Systems



- Encoder: Message → Message Signal or bits
- Transmitter: Message signal → Transmitted signal
- Channel: Introduces noise, distortion, interference
- Receiver: Received Signal → Message Signal
- Decoder: Message Signal → Original Message

Example: Microphone ---------------> Speaker

General Block Diagram of a Communication Transceiver



4- Digital Communication

1 – Orthogonal Frequency Devision Multiplexing (OFDM).
- Introduction to OFDM Systems >>
- Time–Frequency View

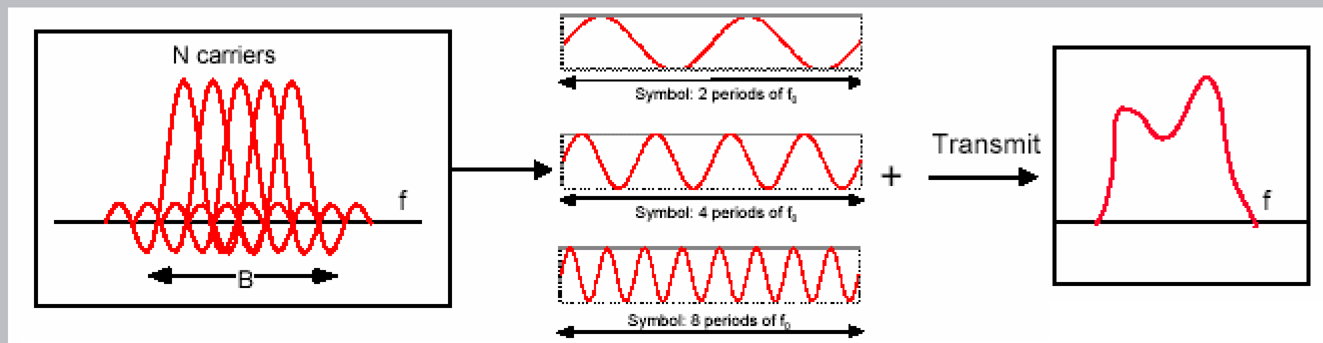
2 – Blind SISO OFDM Channel Estimation Through Pre-Coders and Pre-filters.

3 – Blind MIMO-OFDM Channel Estimation using Independent Component Analaysis. >>

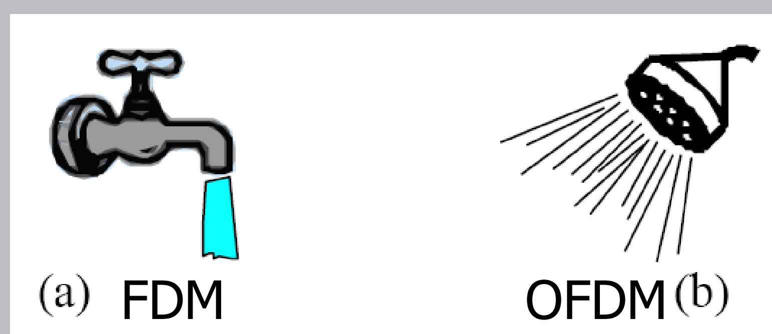
TOP

Introduction to OFDM

OFDM is a multi-carrier communication.



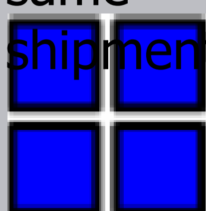
OFDM is a special case of FDM. As an analogy, a FDM channel is like water flow out of a faucet, in contrast the OFDM signal is like shower.



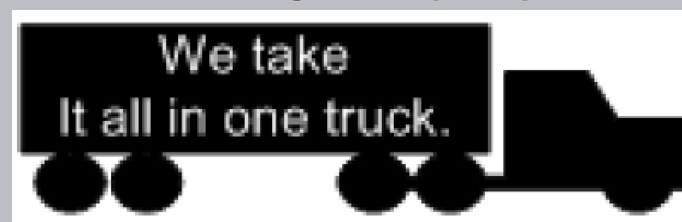
Introduction to OFDM

- Another way to see this intuitively is to use the analogy of making a shipment via truck. Two Options: a big truck or a bounch of smaller one. Both carries the same data. But in the case of an accident, only ¼ of data on OFDM trucking will suffer.

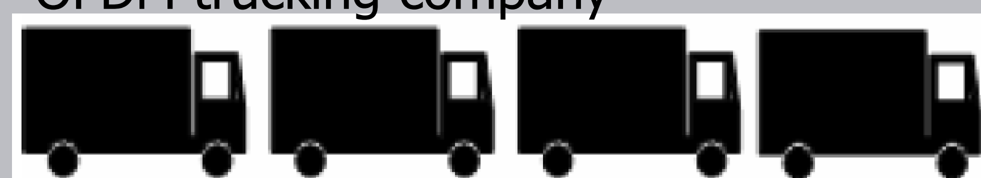
The same shipment



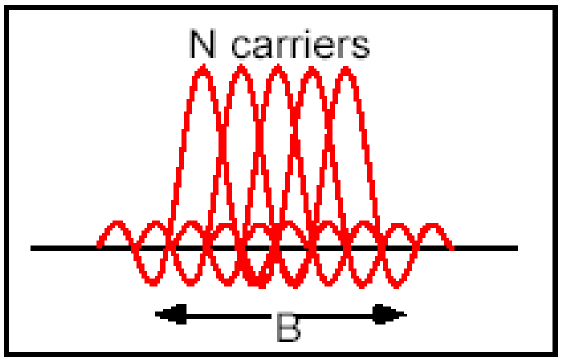
FDM trucking company

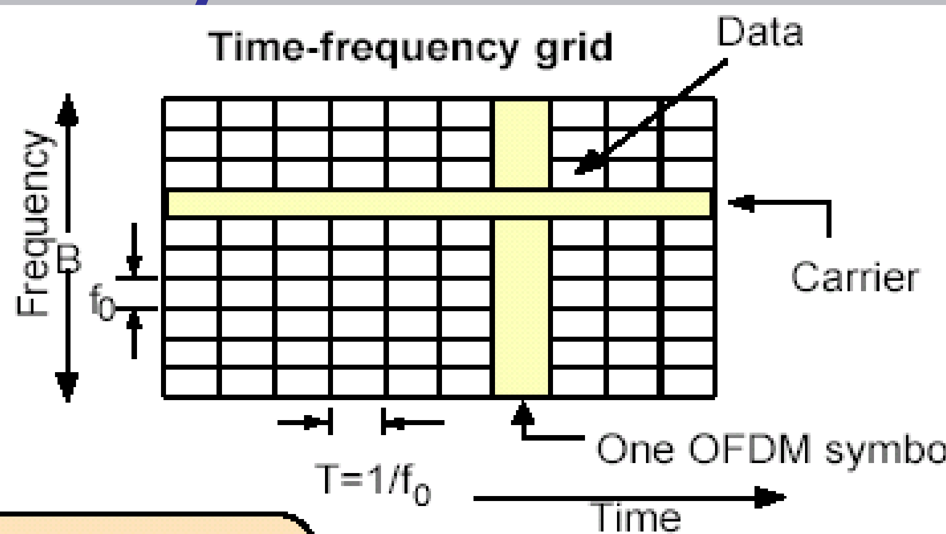


OFDM trucking company



Time-Frequency View





Features

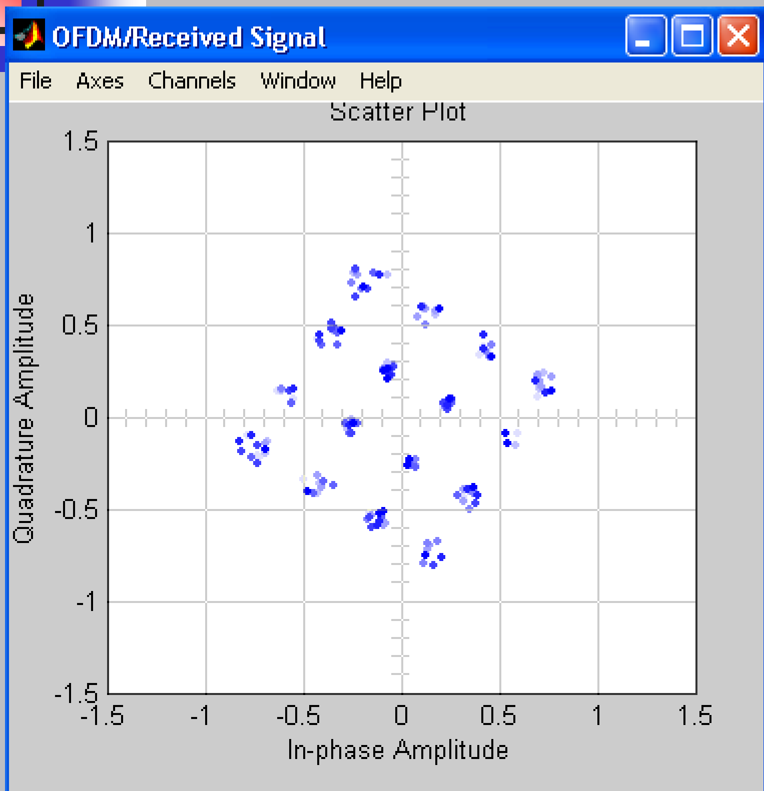
- No intercarrier guard bands
- Controlled overlapping of bands
- Maximum spectral efficiency (Nyquist rate)
- Easy implementation using IFFTs
- Very sensitive to time-freq. synchronization

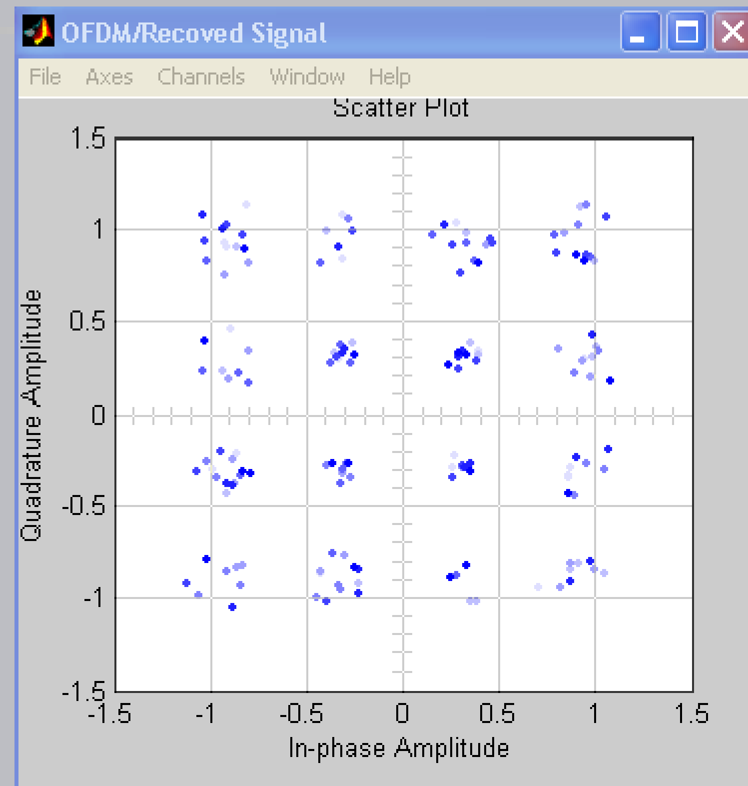
Intercarrier Separation = 1/(symbol duration)

Modulation technique

A user utilizes all carriers to transmit its data as coded quantity at each frequency carrier, which can be quadrature-amplitude modulated (QAM).

Received and Recovered Signals





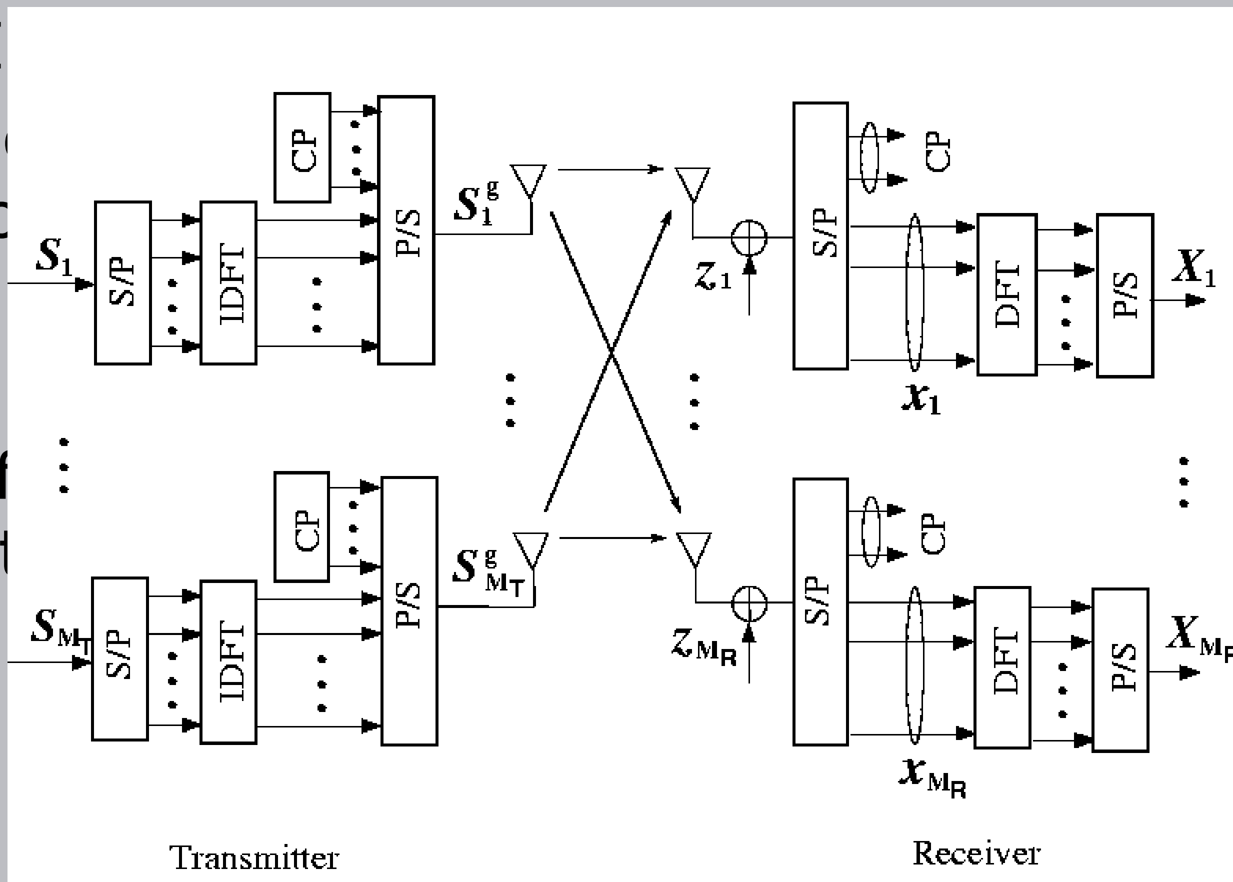
Back

Received signal phases are distorted by multi-path fading

Blind MIMO-OFDM Channel Estimation using ICA

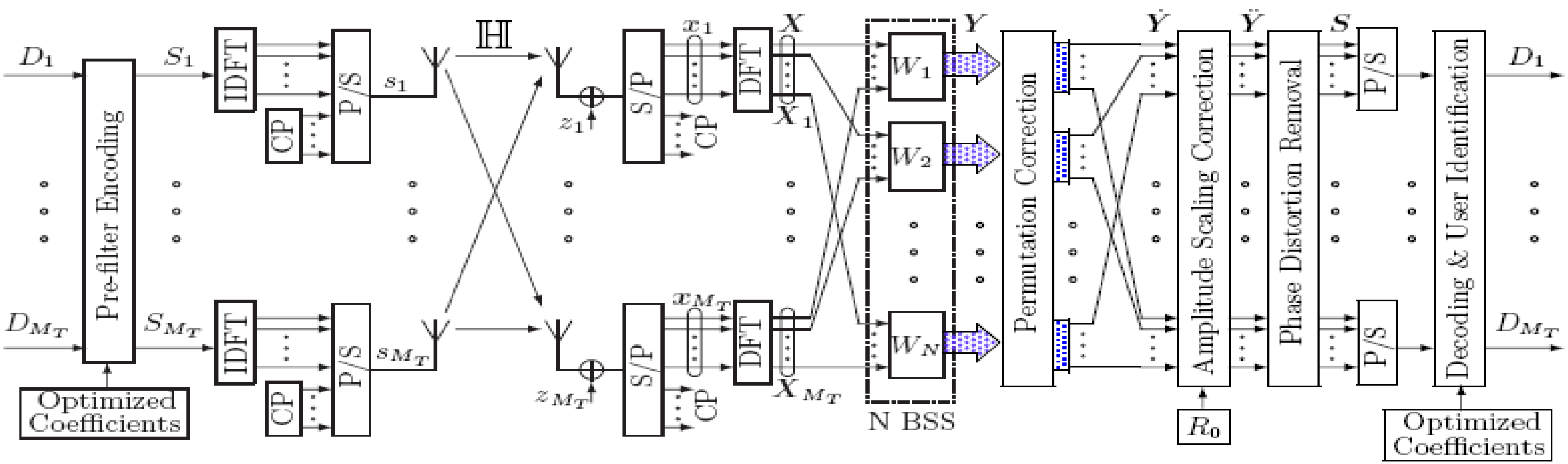
A broadband MIMO-OFDM system model is shown below

- MI
- Th… ome the mo… eme.
- … sible by suf… ser det…



ICA based MIMO OFDM System

In MIMO OFDM system received symbols can be represented as a linear instantaneous mixture of transmitted symbols at each subcarrier m.



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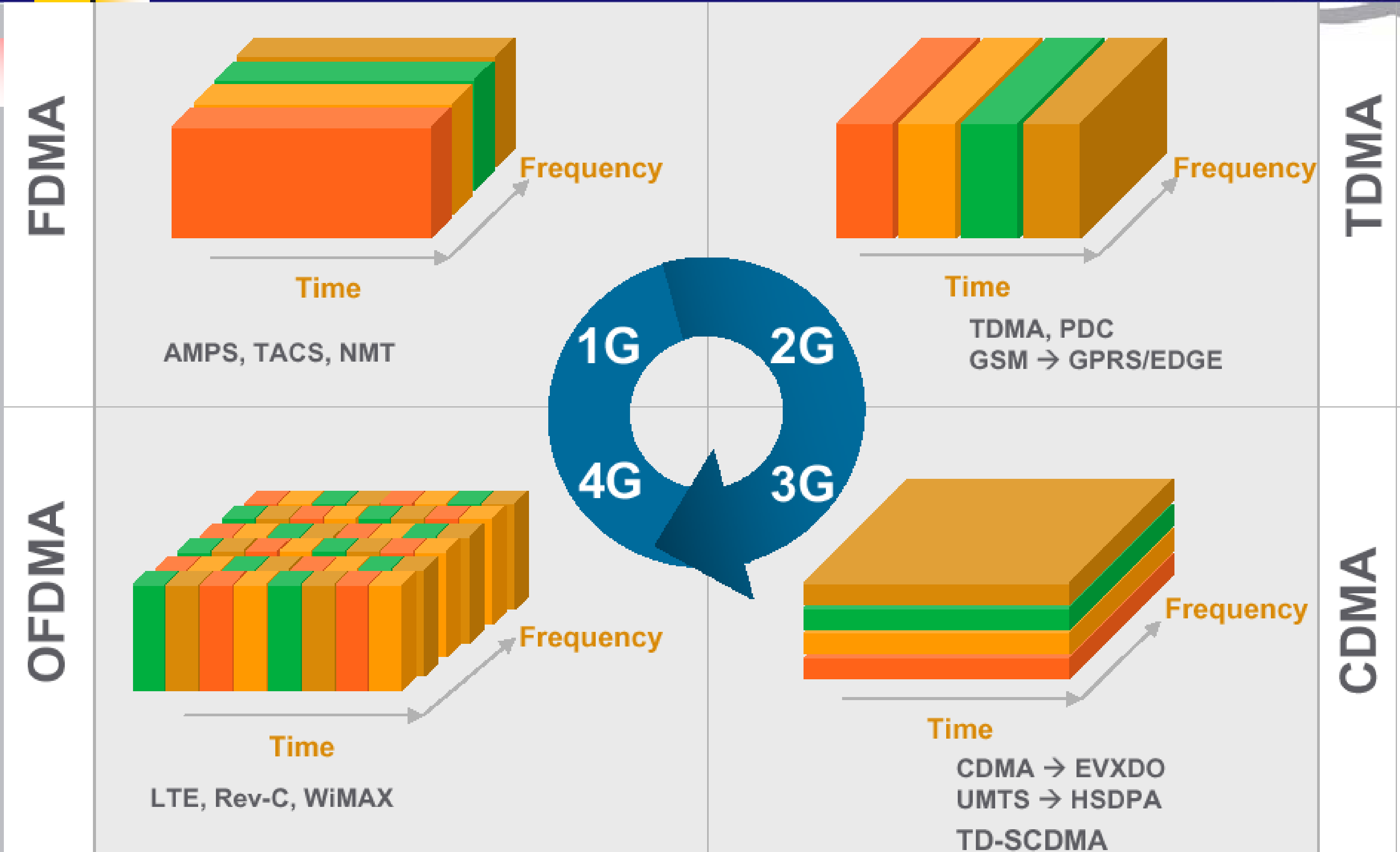
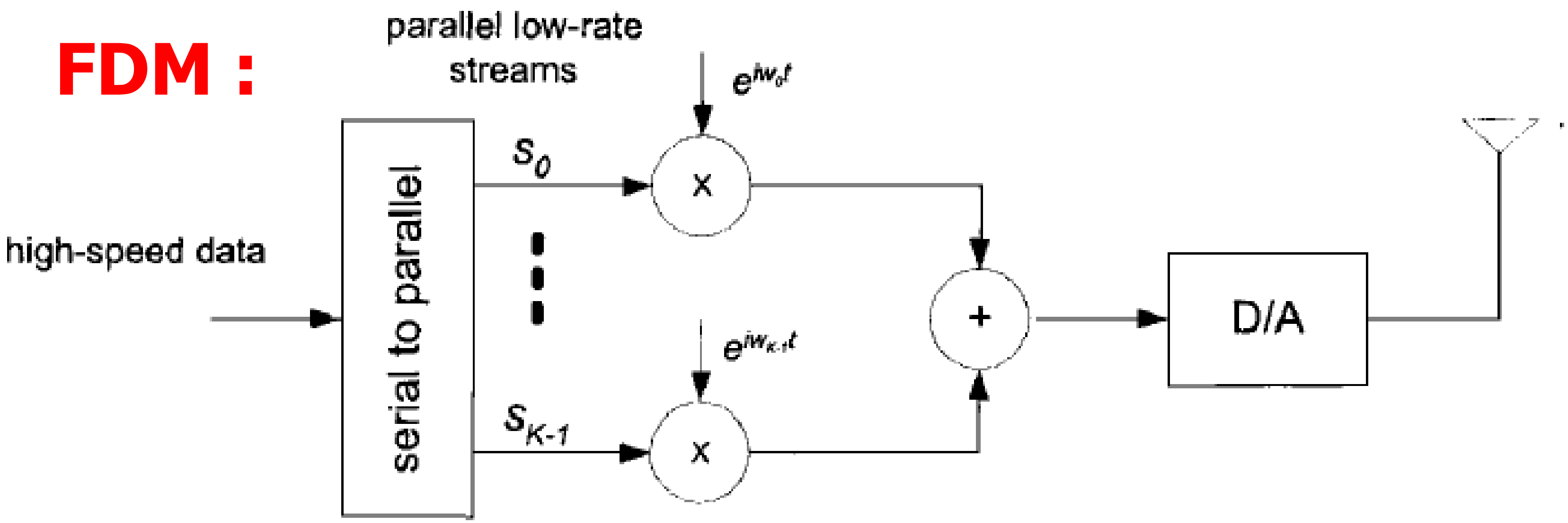
Generations of Digital communication



Comparison between OFDM and FDMA Spectrum

FDM :



FDM scheme for high-speed transmission

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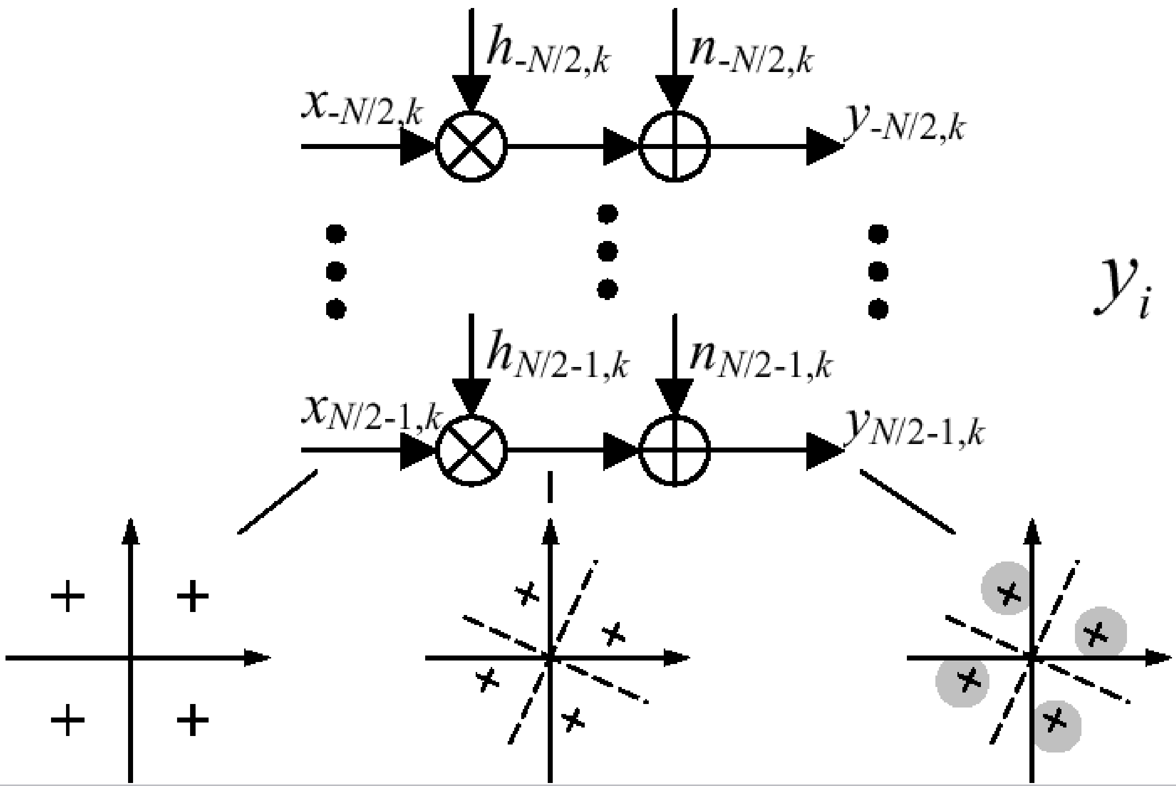
Comparison between OFDM and FDMA Spectrum

OFDM System Model

- Multiplication of data symbols with (complex-valued) channel transfer-function:



$$y_i = x_i h_i + n_i$$

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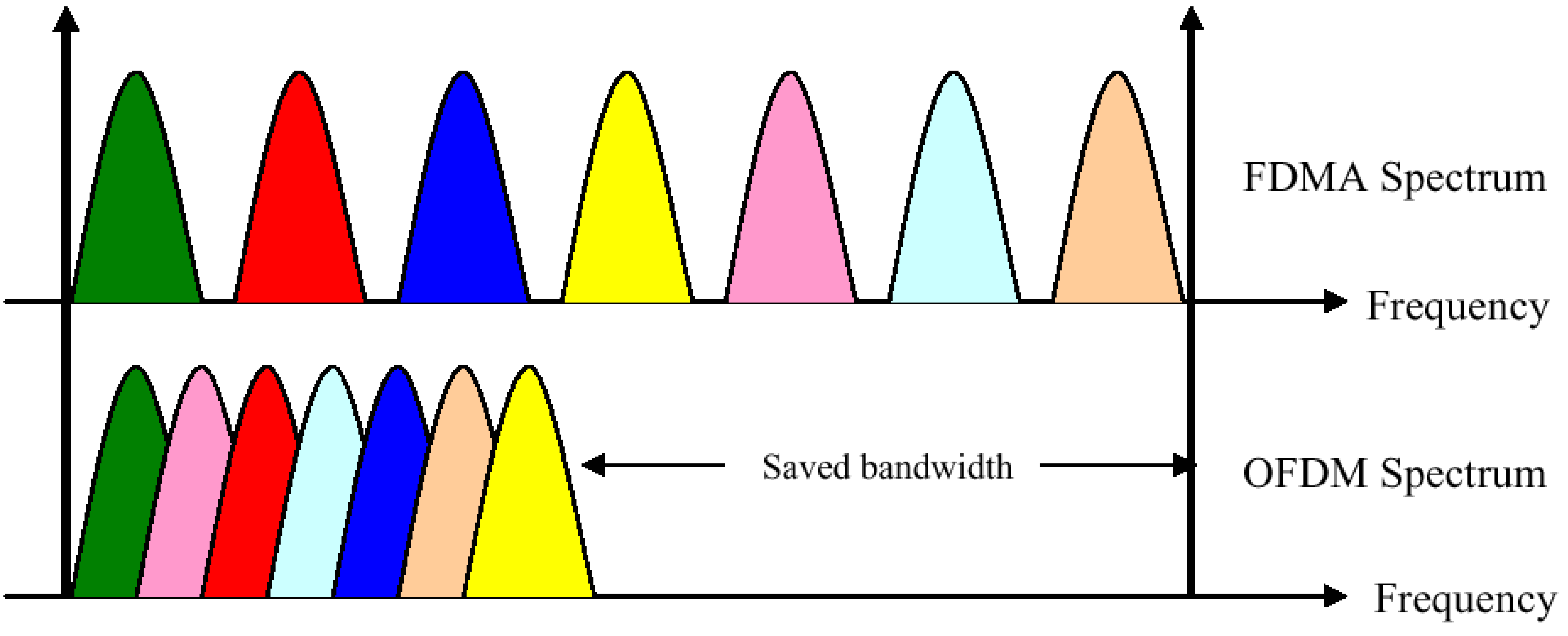
Comparison between OFDM and FDMA Spectrum

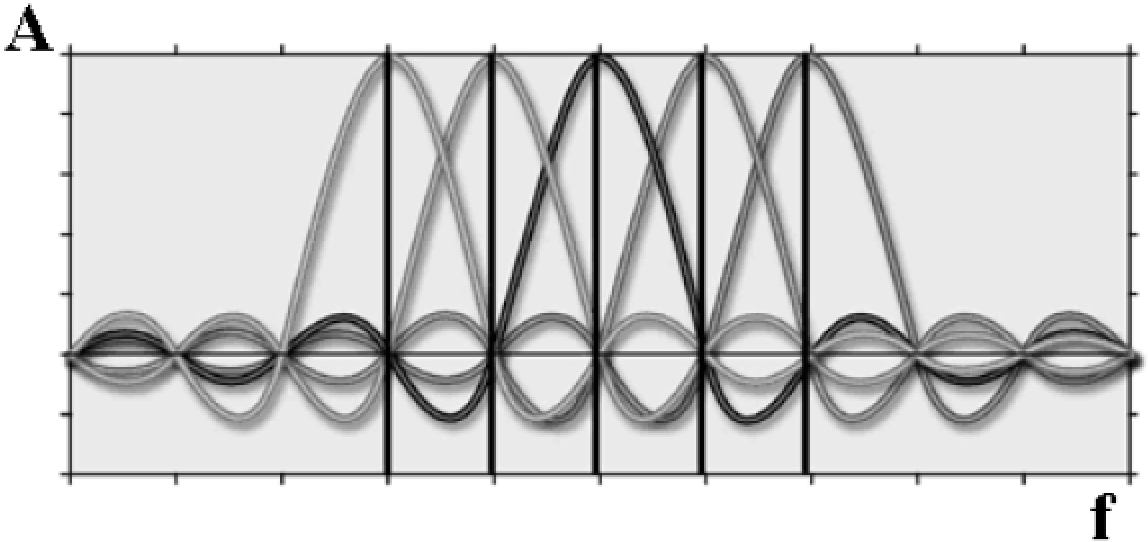


Bandwidth comparison of OFDMA and FDMA

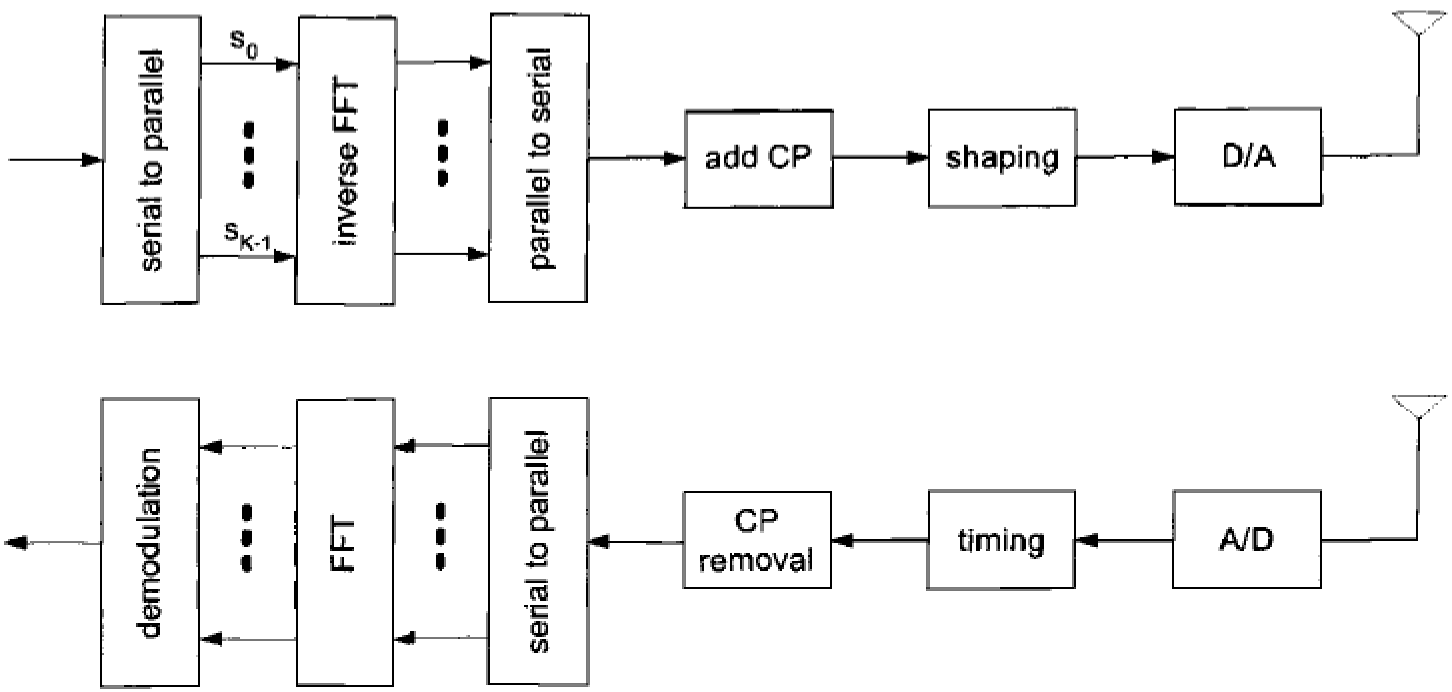
OFDM overview

- Available bandwidth divided into subcarriers
- Subcarriers overlapping but orthogonal with respect to each other

→ at the peak of each subcarrier, the other subcarriers have zero amplitude

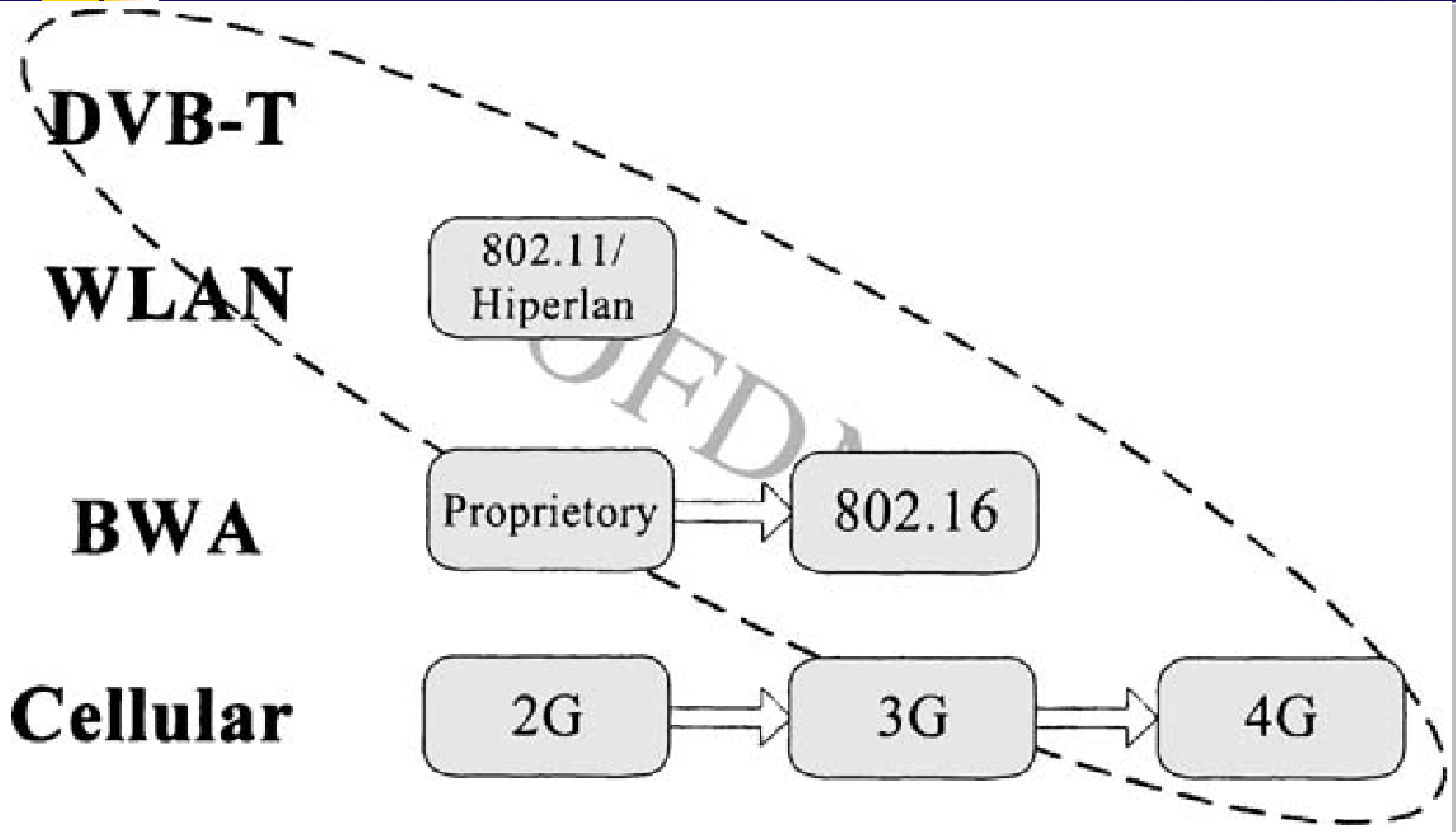


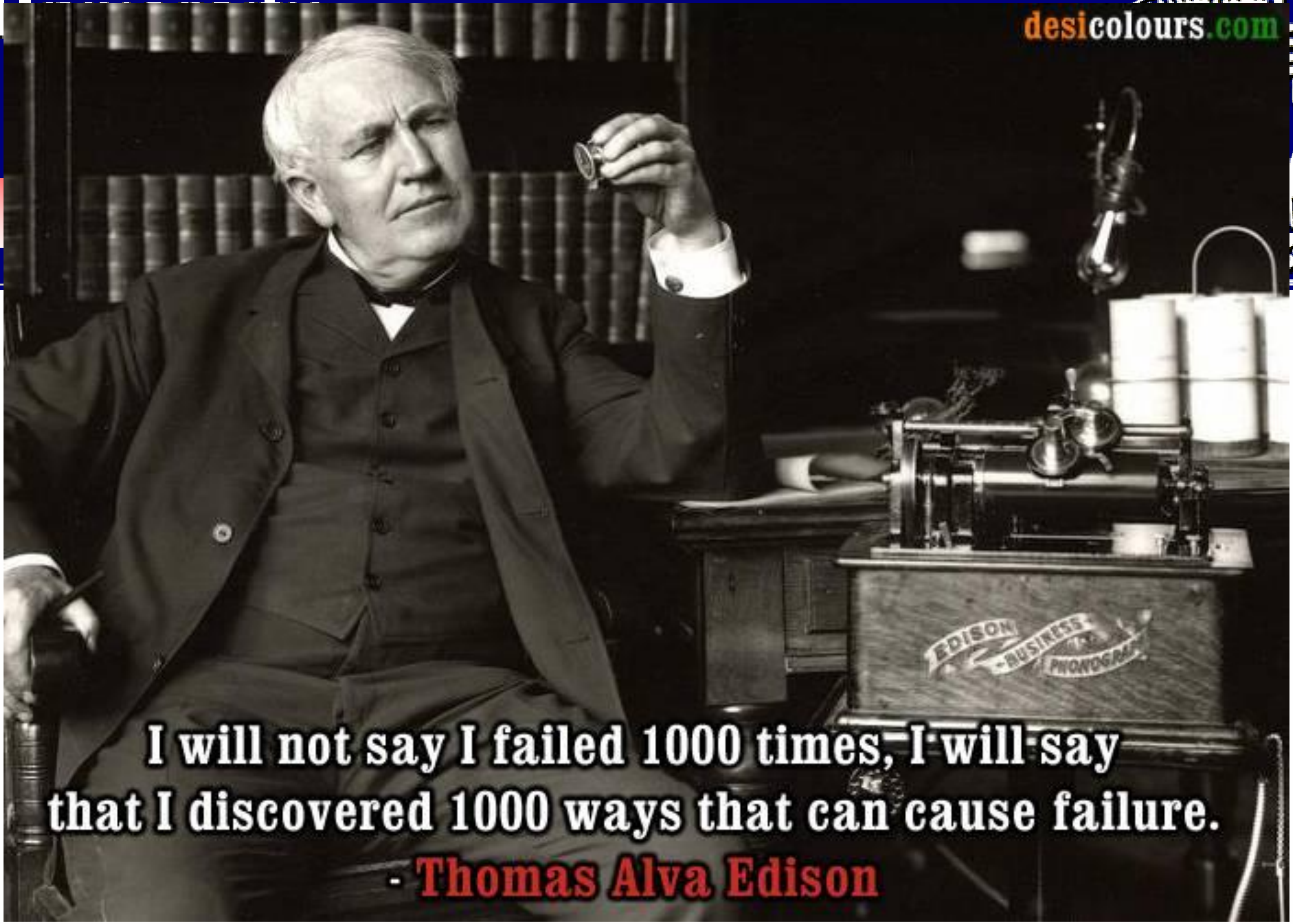
OFDM transceiver block diagram



OFDM transceiver block diagram

Realm of OFDM kingdom

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I will not say I failed 1000 times, I will say
that I discovered 1000 ways that can cause failure.
- Thomas Alva Edison

Thank you!