UNIT – V

Applications of Digital Signal Processing: Introduction, Applications of DSP: Digital Sinusoidal Oscillators, Digital Time Control Circuits, Digital Comb Filters. Applications in broader sense: Removal of noise from pictures, Applications of DSP to Radar, Applications of DSP in Image Processing, Applications of DSP in speech processing.

APPLICATIONS

DSP

Digital Comb Filters.

APPLICATION IN RADAR

APPLICATION TO IMAGE PROCESSING

APPLICATION TO SPEECH PROCESSING

Digital Sinusoidal Oscillators,

Digital Time Control Circuits,

Removal of noise from pictures,

APPLICATION TO IMAGE PROCESSING

Manipulation of two-dimensional signal with the help of digital computer is called "Image Processing". Its purpose is to improve the visual appearance of image.

A Digital Image is digitalization of picture. Normally two-dimensional image has resolution 128×128, 256×256, 512×512. So, image can be processed using two-dimensional signal processing.

The image processing includes the following steps :

(a) Image Formation and Recording
(b) Image Sampling and Quantization
(c) Image Compression
(d) Image Restoration
(e) Image Enhancement

(a) Image Formation and Recording

The two dimensional signal of image can be expressed by image function as

$$g(x, y) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} h(x - x_1, y - y_1) f(x_1, y_1)\, dx_1\, dy_1$$

point-spread function,

accumulation of energy from the object's radiant energy distribution.

Two major technologies are used for image sensing and recording,

photo-chemical recording and photo-electronic recording.

(b) Image Sampling and Quantization

a spot of light with intensity I_1 incident on a film

intensity I_2 **REFLECTED** from the film and collected by photo-multiplier.

transmittance is defined by $T = \frac{I_2}{I_1}$ compute optical density.

a spot of light, moves in a raster to sample the film is given by

$$g_1(x, y) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} h_a(x - x_1, y - y_1)\, g(x_1, y_1)\, dx_1\, dy_1$$

intensity profile of the spot of light projected an film.

g_1 is actual sampled image.

The sample matrix $g_1(k\,\Delta x, l\,\Delta y)$ is the *sampled* **or DIGITAL IMAGES**

(c) Image Compression

In a Digital Image 10^5 to 10^6 data are there.

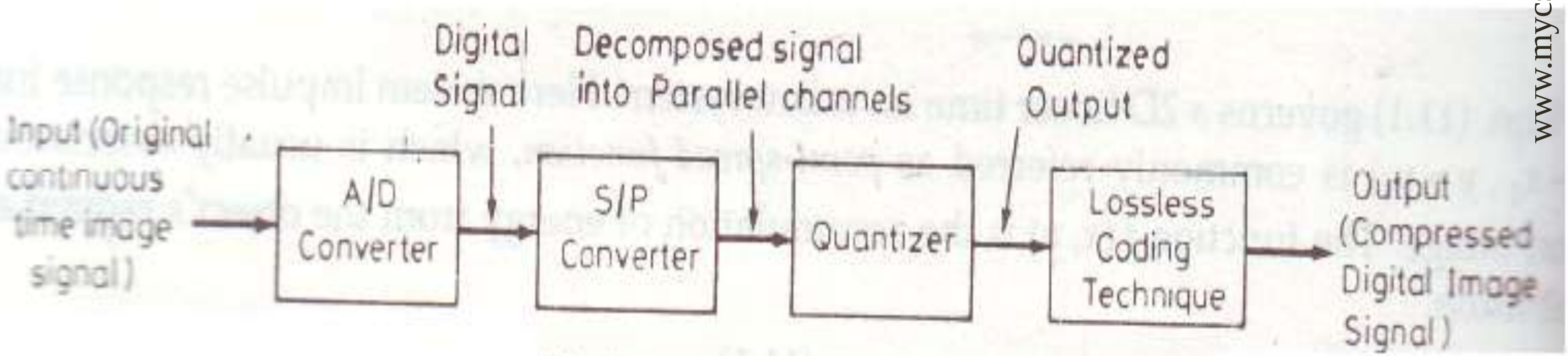
Image compression is a science of efficiently coding a digital image to reduce the number of bits, which required to represent it.

Types of redundancy: correlation neighbouring pixels.

a) Spatial redundancy,

b)Spectral redundancy, Correlation between various color plans

c)Temporal redundancy. correlation between differen**frams in an image sequence**



Block diagram of image compression system.

There are mainly three types of compression technique based on the method of redundancy detection :

(a) Direct data compression method.

(b) Transformation method.

(c) Parametric extraction method.

(d) Image Restoration

- ❖ It is used for correcting imaging effect to recover an original signal.
- ❖ The process of image restoration is to attempt a image which should be sharp, clean and free from degradation.
- ❖ The restoration process is also called Image Deblurring.

R is the response characteristic of the recording process

$$g(x, y) = R\left[\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h(x - x_1, y - y_1) f(x_1, y_1)\, dx_1\, dy_1\right] + n(x, y)$$

g (x, y) is the actial image,

additive noise source.

In the restoration of digital image following equation can be expressed in discrete form :

$$g(p,q) = \sum_{i=0}^{N-1} \sum_{j=0}^{N-1} f(i,j)\, h(p-i, q-j)$$

large set of simultaneous linear equations can be solved by DSP techniques

such as linear filters

FFT algorithms which are computationally efficient tools for solving these.

(e) Image Enhancement

- This technique improves the appearance of image for human perception by choosing some image features like edges or contrast etc.
- It is used in biomedical engineering field for computer aided mammographic studies.
- Finger prints etc.

APPLICATION IN RADAR

- RADAR is an acronym for **Radio Detection and Ranging.**
- RADAR system requires seven basic components

1. Transmitter
2. Receiver
3. Power supply
4. Synchronizer
5. Duplexer
6. Antenna
7. Display

- Radar is used to detect stationary and moving objects.

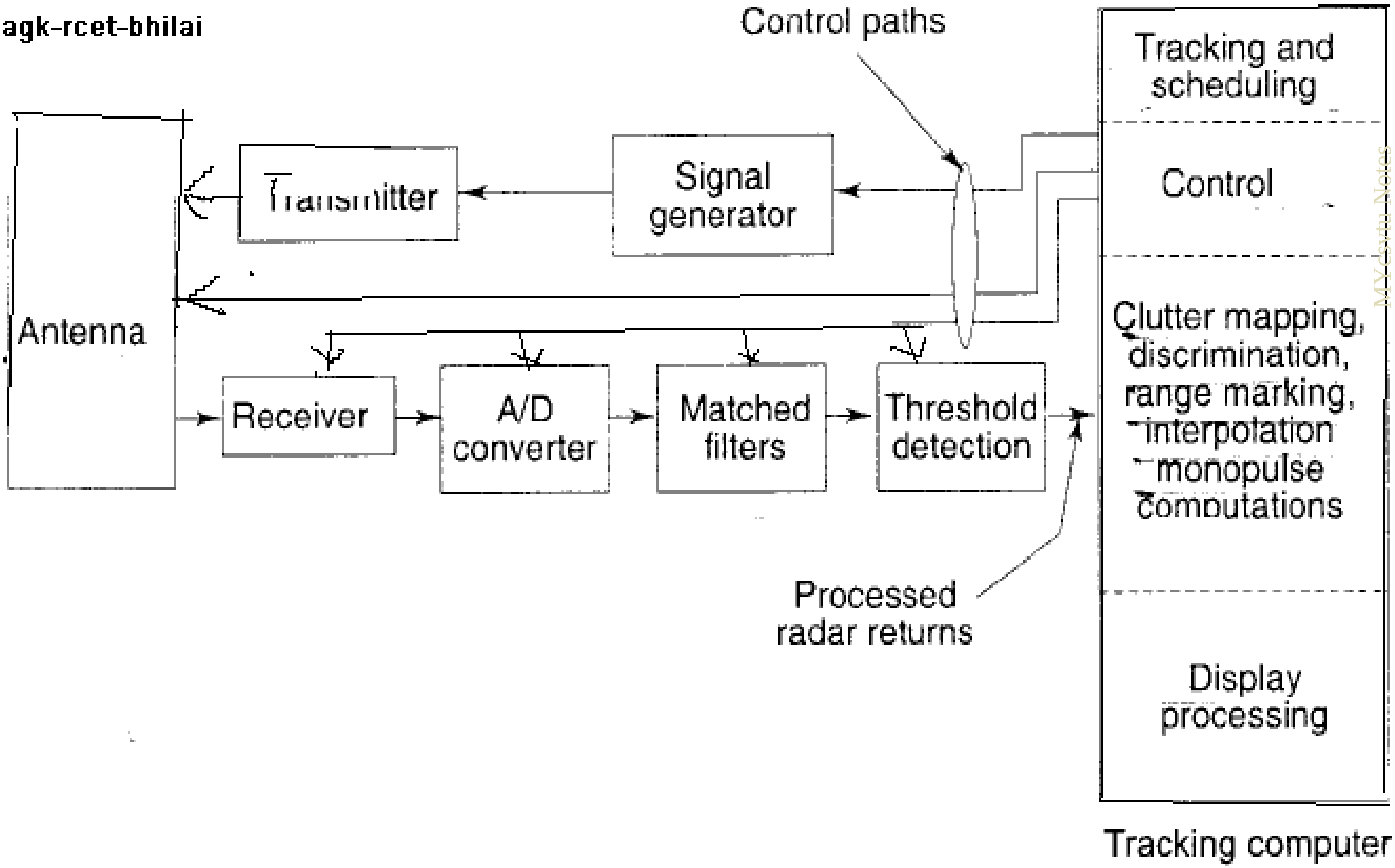
- Major components are Antenna(6), the tracking computer and signal processor.
- Heart of the RADAR is tracking computer which schedules the appropriate antenna position and transmitted signals which is a function of time by keeping track of important targets and running the display system(7).
- The main functions of signal processor are matched filtering and removal of useless information by threshold detection(4).
- IEEE standard RADAR frequencies are

  VHF:30-300MHz

  UHF:300-1000MHz

Block Diagram of a Modern Radar System

agk-rcet-bhilai



important radar parameters are

[1] ***Antenna Beamwidth***

$$\beta \propto \frac{\lambda}{D}$$

where

β - beamwidth

λ - wavelength

D - antenna width

For a pencil beam, the antenna geometry is symmetric, β is the same in both the horizontal and vertical dimensions.

[2] **Range** The maximum unambiguous range is

$$R_{\max} = \frac{cT}{2}$$

where

c - velocity of light = 3×10^8 m/s

T - pulse repetition interval

[3] Range Resolution

If two targets are present near each other, then the ability of the radar to detect these targets are measured by the range resolution ΔR. If the signal is of constant frequency, then ΔR is determined by the pulse width. If the pulse width is narrowed, then range resolution can be improved but the maximum range is reduced by decreasing the average power.

[4] Doppler Filtering

Moving targets can be identified by using the Doppler effect.

if we get received echo-signal as $f_o + \Delta f$, then

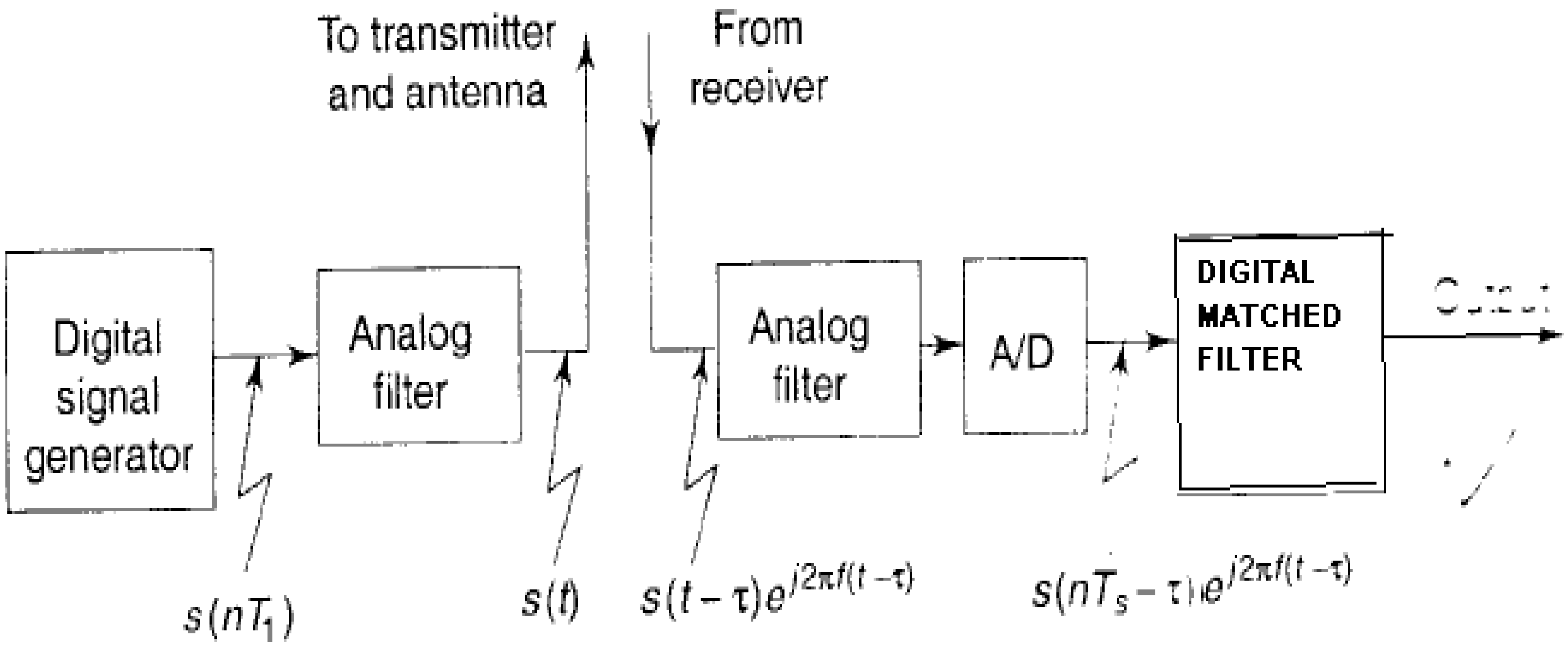
$$\Delta f = \frac{2v}{c} f_0$$

$$= \frac{2v}{\lambda}$$

λ - wavelength

where f_0 - carrier frequency
v - target velocity

Block Diagram of a Radar Model



Example 01:
Calculate the range of a target if the time taken by the radar signal to target and back is 100 μs.

Solution: R= 15km

Example 02:
Determine the maximum unambiguous rang and range resolution of a pulse radar having pulse width is 5 μs at a PRF of 1000 Hz.
Solution:
Pulse repetition frequency……………fr
Rmax(unamb)=150km -----------Rmax=c/2fr
Range resolution=750m

Example 03:

A radar is to have a maximum range of 250 km .Determine the maximum allowable PRF for unambiguous reception.

Solution:

Fr=600hz.

Example 04:

The maximum unambiguous range of 1000 km, calculate the PRF required for the radar.

Solution:

1500 km.

APPLICATION TO SPEECH PROCESSING

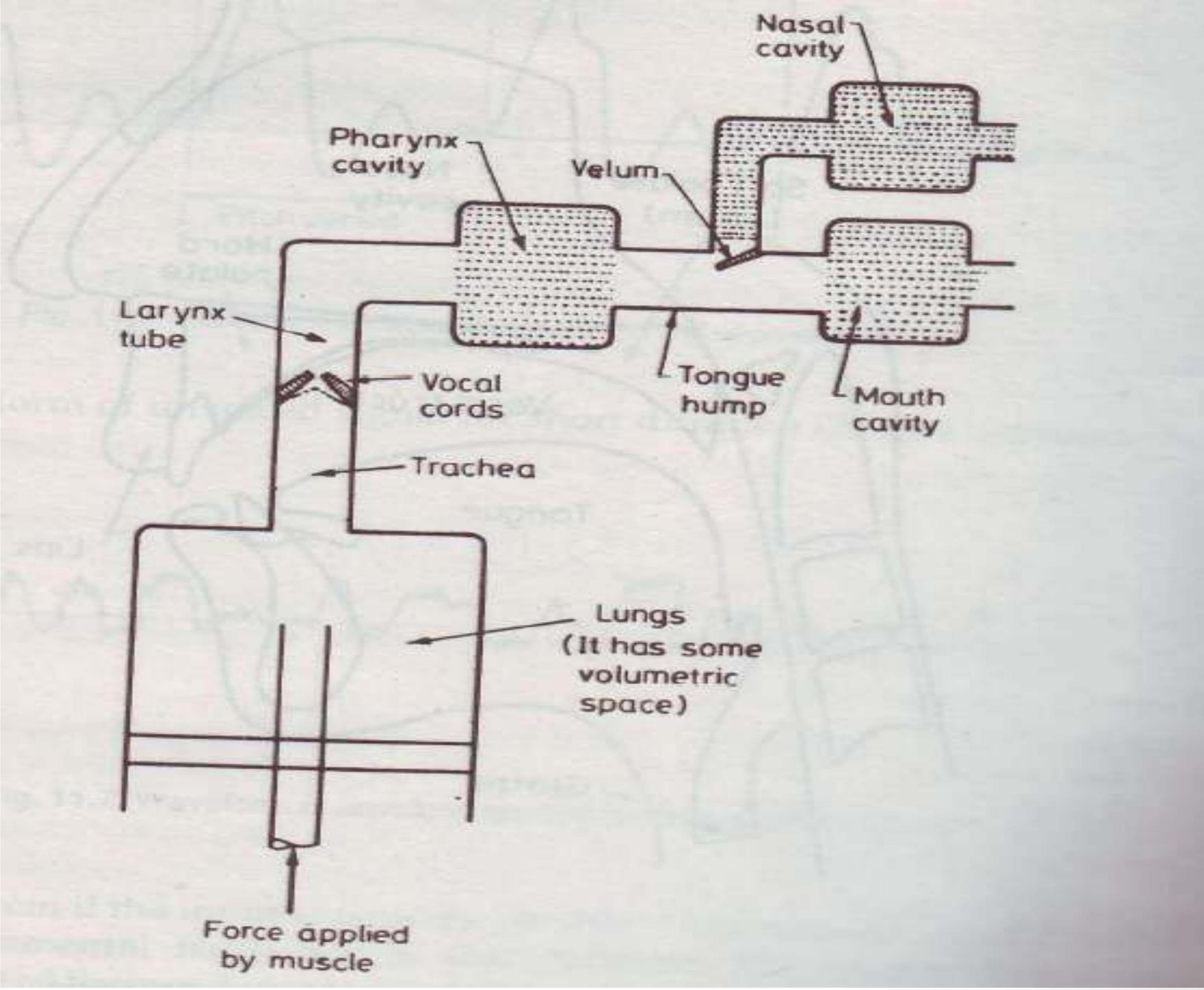
- The signals of speech are one dimensional.
- DSP is applied to speech such as vocoders, spectrum analysis etc.
- Speech processing can be divided in to three categories:

(1)Speech analysis,
(2) Speech synthesis,
(3) Speech compression.

Model of Speech Production

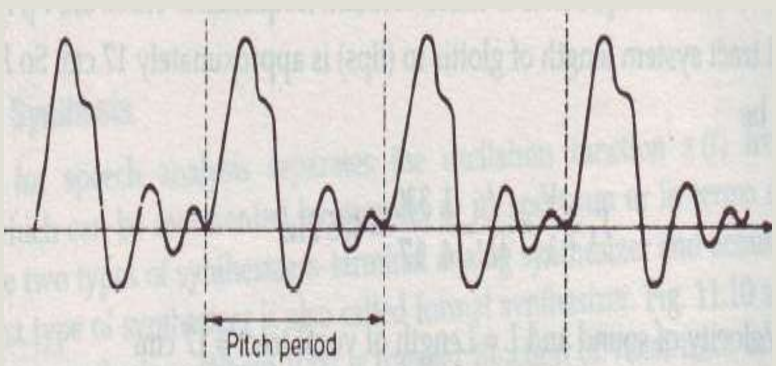
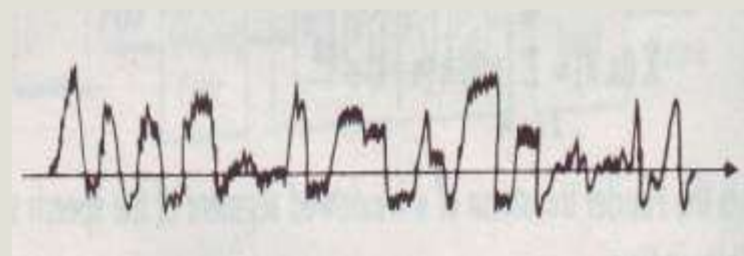
In normal speech production of human speech production system. The chest cavity expands and contacts to force air from the lungs out through the trachea past the glottis.
The mechanism is:

Schematic diagram of human speech production mechanism.

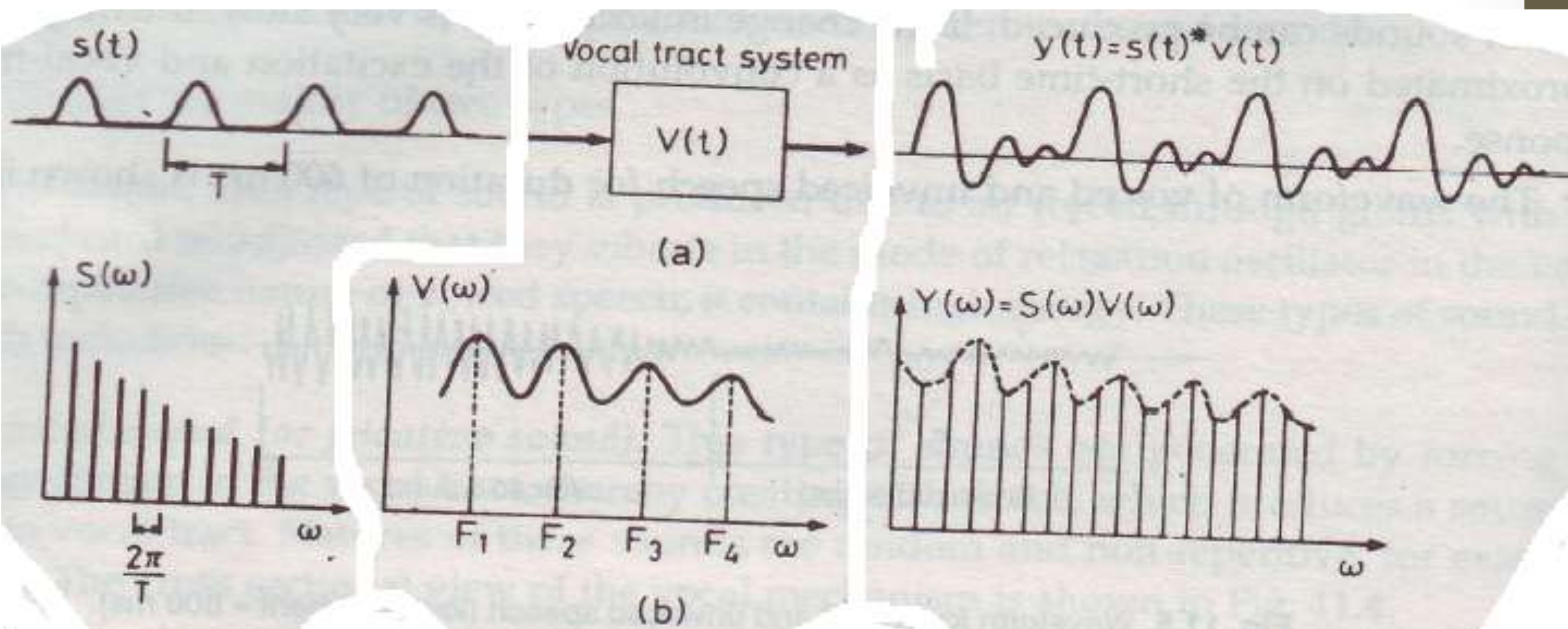


Nasal cavity
Pharynx cavity
Velum
Larynx tube
Vocal cords
Tongue hump
Mouth cavity
Trachea
Lungs (It has some volumetric space)
Force applied by muscle

There are two types of speech sound,

	[1]Voiced sound	[2]Unvoiced sound
	Rich melodies This type of sound is produced due to air forced through glottis with the tension of vocal card so adjusted that they vibrate in the mode of relaxation oscillator in the vocal tracts. Due to repetitive nature of voiced speech, it contains high energy.  Waveform for voiced speech (short segment = 25 ms).	Hissing sound This type of sounds are generated by forcing air through a constriction in the vocal tract, thereby creating turbulence, which produces a source of noise to excite vocal tract. Natures of these sounds are random and non-repetitive,  Waveform of unvoiced speech (short segment = 25 ms).

In a vocal tract system if the input is periodic on short time basis for voiced speech the output to corresponding fundamental frequency is also periodic. **The Fourier Transform of speech waveform is the product of Fourier Transform of excitation function and impulse response of vocal tract system.**



(a) Time domain and (b) Frequency domain characterization of vocal tract system.

Vocal tract resonant frequencies is called formant frequencies. These are F_1, F_2, F_3 and so on. In human vocal tract system length of glottis to (lips) is approximately 17 cm. So F_1 (fundamental frequency) will be

$$F_1 = \frac{V}{\lambda} = \frac{V}{4L} = \frac{1}{4}\frac{330}{17} = 485 \text{ Hz.}$$

Here V = Velocity of sound and L = Length of vocal tract = 17 cm.

Short-Time Fourier Transform and Synthesis of Speech

If we consider speech signal, to be sampled, is a sequence $x(n)$ then the short-time transform $X(\omega, n)$ is defined as

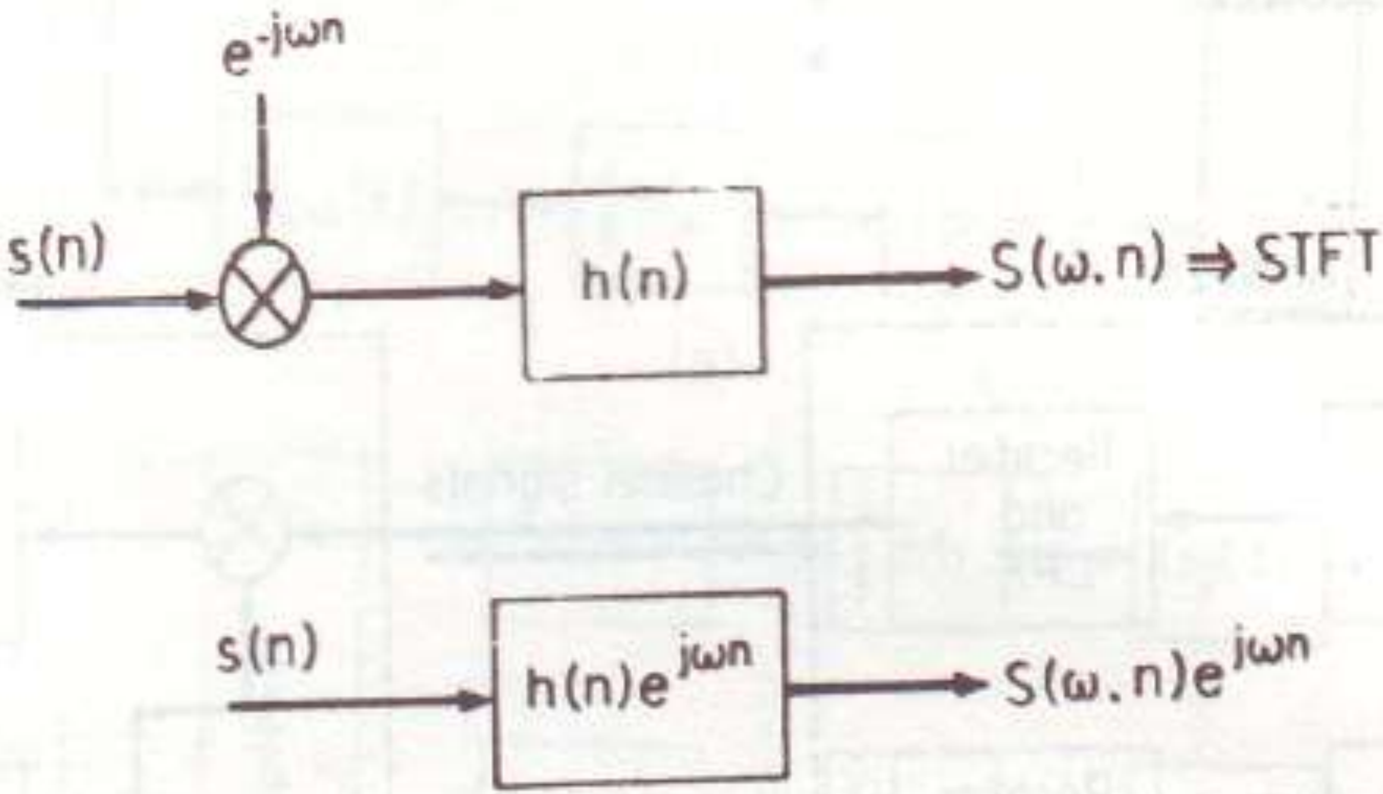
$$X(\omega, n) = \sum_{k=-\infty}^{\infty} x(k)\, h(n-k)\, e^{-j\omega k}$$

To implement STFT, if we have $h(n)$ is impulse response of vocal tract and $x(n)$ is input signal. The convolution of $x(n)$ and $h(n)$ multiplied by $e^{-j\omega k}$ gives STFT *i.e.*,

$$X(\omega, n) = \sum_{k=-\infty}^{\infty} x(k)\, h(n-k)\, e^{-j\omega k}$$

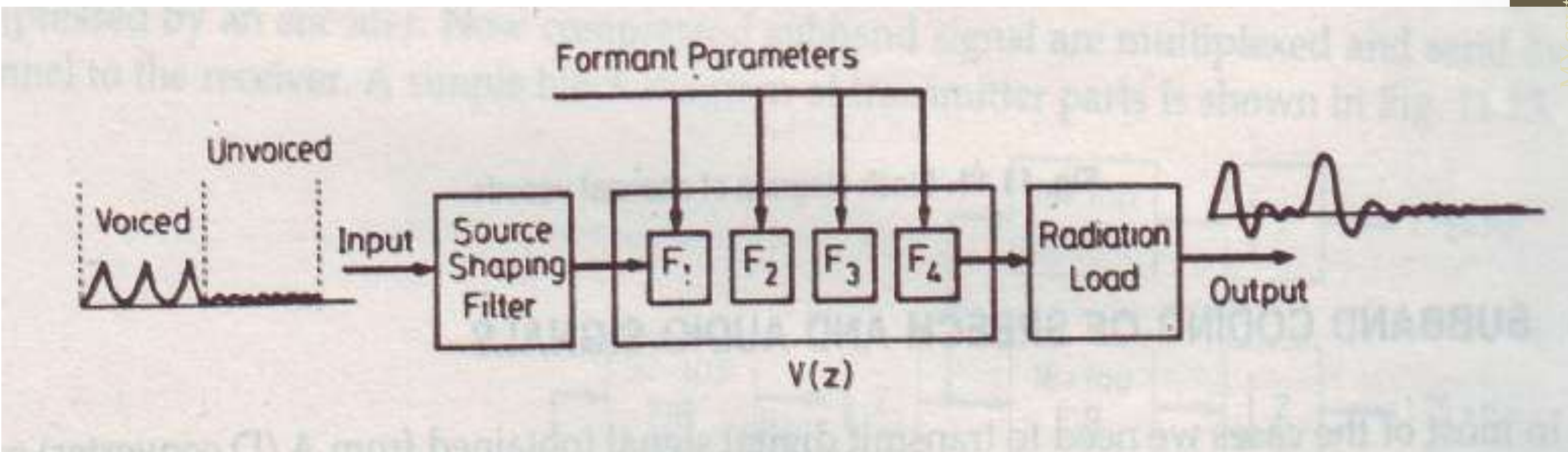
$$X(\omega, n) = e^{-j\omega n}\{x(n) * [h(n)\, e^{j\omega n}]\}$$

We can represent this Short-Time Fourier Transform (STFT) in block diagram



Speech Synthesis

A system for speech analysis separates the excitation function $x(t)$ from vocal tract characteristic which can be represented by samples of its spectrum or in terms of a parametric



Block Diagram of format synthesizer.

DIGITAL SINUSOIDAL OSCILLATOR

INTRODUCTION: A digital resonator is a special two pole band pass filter with pair of complex conjugate poles located near the unit circle. It is called resonator because the filter has large magnitude response (i.e. it resonates) near the vicinity of pole location. Digital resonators are useful in many applications including simple band pass filtering & speech generation. A digital sinusoidal oscillator is a limiting form of a two-pole resonator for which the complex-conjugate poles lie on the unit circle.

TRASFER FUNCTION: The system has following transfer function

$$H(z) = \frac{b_0}{1+a_1 z^{-1}+a_2 z^{-2}} \qquad \text{------------------------ (1)}$$

and parameters $a_1 = -2r\cos\omega_0$ and $a_2 = r^2$

has complex-conjugate poles at $p = r\,e^{\pm j\omega 0}$, and a unit sample response

$$h(n) = \frac{b_0 r^n}{\sin\omega_0}\sin(n+1)\omega_0\, u(n)$$

has complex-conjugate poles at $p = r e^{\pm j\omega_0}$, and a unit sample response

$$h(n) = \frac{b_0 r^n}{\sin \omega_0} \sin(n+1)\omega_0 \, u(n)$$

If the poles are placed on the unit circle (r=1) and b_0 is set to A sin ω_0, then

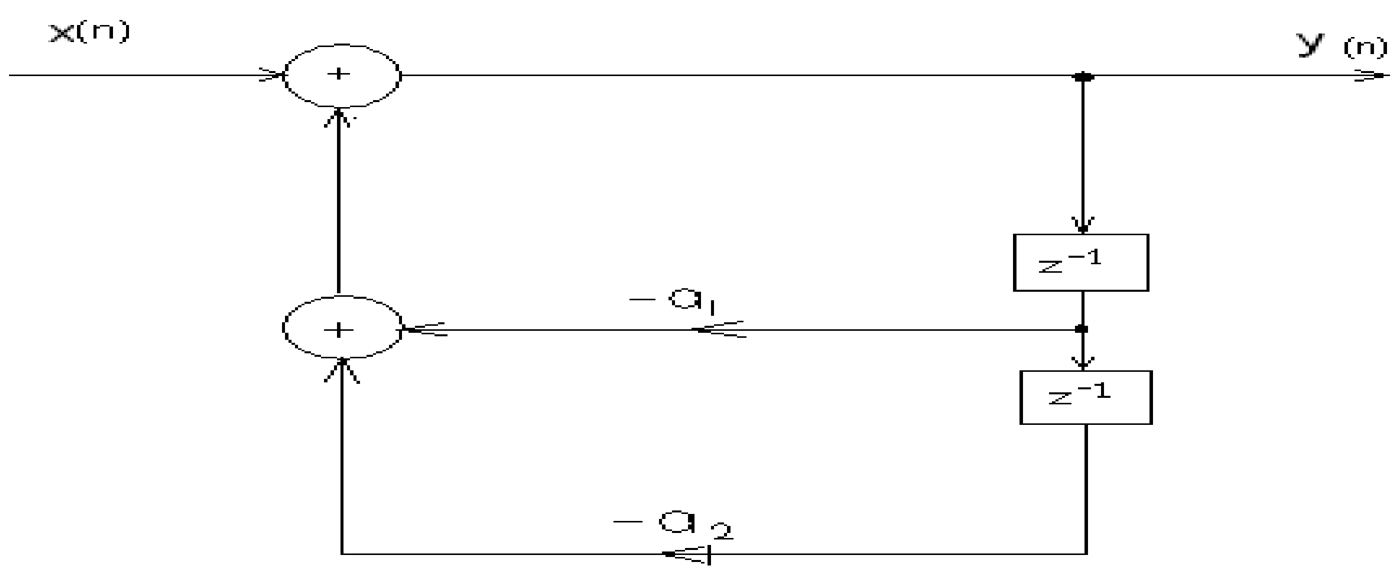
$$h(n) = A \sin (n+1)\omega_0 u(n)$$

Thus the impulse response of the second-order system with complex-conjugate poles on the unit circle is a sinusoid and the system is called a digital sinusoidal oscillator or a digital sinusoidal generator. A digital sinusoidal generator is a basic component of a digital frequency synthesizer.

BLOCK DIAGRAM REALIZATION: The block diagram realization of system function given by equation (1) is shown in figure below. The corresponding difference equation for this system is

$y(n)=-a_1\, y(n-1) - y(n-2) + b_0\delta(n)$ ---------------------- (2)

where the parameters are $a_1=-2\cos\omega_0$ and $b_0= A\sin\omega_0$,and the initial condition are $y(-1)=y(-2)=0$.



By iterating the equation we obtain

$y(0)=A \sin \omega_0$

$y(1)=2 \cos \omega_0\, y(n)=2A \sin \omega_0 \cos \omega_0=A \sin 2\omega_0$

$y(2)= 2 \cos \omega_0\, y(1)-y(0)$

$=2A \cos \omega_0 \sin \omega_0-A \sin \omega_0$

$=A(4 \cos^2 \omega_0 -1)\sin \omega_0$

$=3A \sin \omega_0-4 \sin^3 \omega_0=A \sin 3\omega_0$

And so forth. We note that the application of the impulse at $n=0$ serves the purpose of beginning the sinusoidal oscillation. Thereafter, the oscillation is self-sustained because the system has nc damping (i.e. $r=1$).

The sinusoidal oscillation obtained from the system is also be obtained by setting the input tc zero and setting the init ial condition to $y(-1)=0, y(-2)=-A \sin \omega_0$.Thus the zero-input response tc the second-order system described by the homogeneous difference equation

$y(n) = -a_1 y(n-1) - y(n-2)$

with initial conditions $y(-1)=0$ and $y(-2) = -A\sin\omega_0$, is exactly the same as the response of equation(2) to an impulse excitation.

COUPLED FORM OSSCILATOR REALIZATION: In some practical applications involving modulation of sinusoidal carrier signals in phase quadrature, there is a need to generate the sinusoids $A\sin\omega_0 n$ and $A\cos\omega_0 n$. These signals can be generated from the coupled-form oscillator, which can be obtained from the trigonometric formulas

$\cos(\alpha+\beta)=\cos\alpha\cos\beta-\sin\alpha\sin\beta$

$\sin(\alpha+\beta)=\sin\alpha\cos\beta+\cos\alpha\sin\beta$

Where, by definition, $\alpha=n\omega_0$, $\beta=\omega_0$, and

$y_c(n)=\cos n\omega_0 u(n)$

$y_s(n)=\sin n\omega_0 u(n)$

Thus we obtain the two coupled difference equations

$y_c(n)=(\cos\omega_0)y_c(n-1)-(\sin\omega_0)y_s(n-1)$

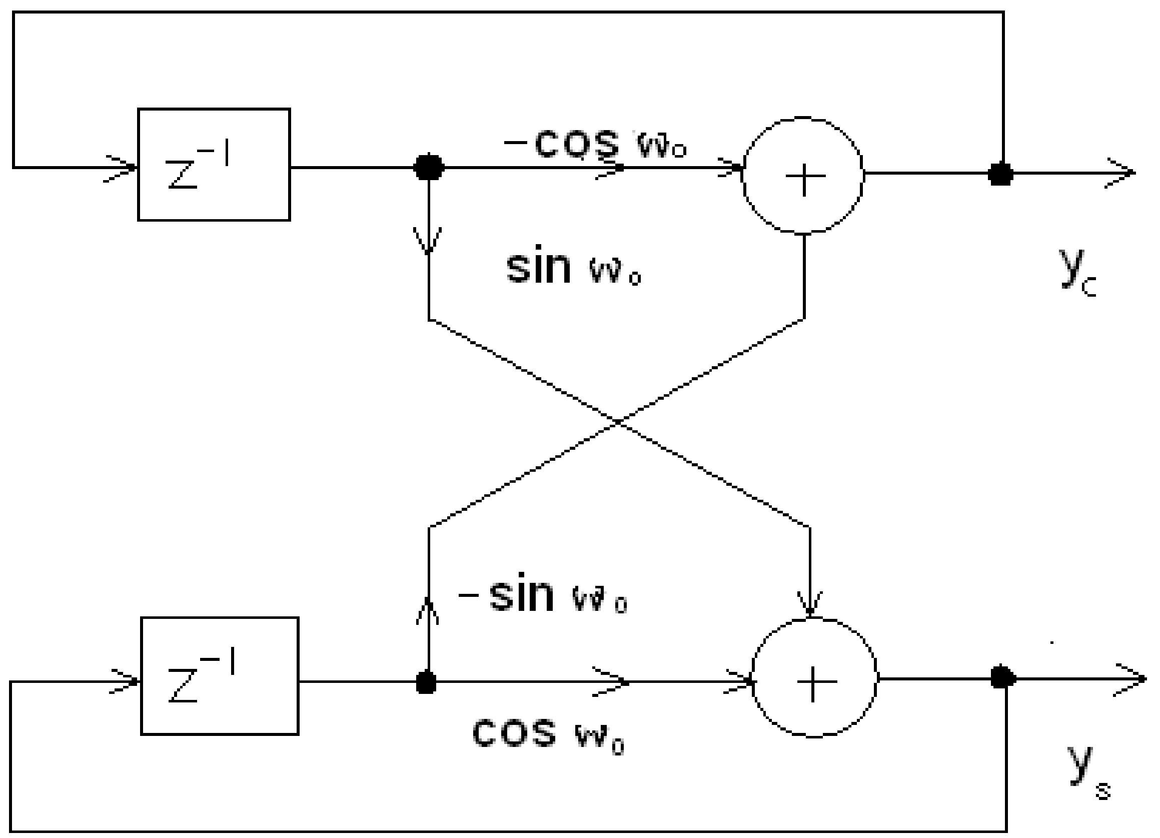
$y_s(n)=(\sin\omega_0)y_c(n-1)+(\cos\omega_0)y_s(n-1)$

Which can also be expressed in matrix form as:

$$\begin{vmatrix} y_{c(n)} \\ y_{s(n)} \end{vmatrix} = \begin{matrix} \cos\omega_0 & -\sin\omega_0 \\ \sin\omega_0 & \cos\omega_0 \end{matrix} * \begin{matrix} y_{c(n-1)} \\ y_{s(n-1)} \end{matrix}$$

The structure for the realization of the coupled-form oscillator is illustrated in figure. We note that this is a two-output system which is not driven by any input, but which requires the initial conditions $y_c(-1)=A\cos\omega_0$ and $y_s(-1)=A\sin\omega_0$ in order to begin its self-sustained oscillations.

The corresponds to vector rotation in the two-dimensional coordinate system with coordinates $y_c(n)$ and $y_s(n)$. As a consequence, the coupled-form oscillator can be implemented by use of the so-called CORDIC algorithm .



z^{-1}
$-\cos w_0$
$+$
$\sin w_0$
y_c
$-\sin w_0$
z^{-1}
$+$
$\cos w_0$
y_s

Noise removal from Pictures

Introduction: **Digital images are prone to a variety of types of noise. Noise is the result of errors in the image acquisition process that result in pixel valves that do not reflect the true intensities of the real scene. Noise can be introduced into an image, depending on how image is created.**

For example:

If the image is scanned from a photograph made on film, the film grain is a source of noise. Noise can also be the result of damage the film, or be introduced by the scanner itself.

If the image is acquired directly in a digital format, the mechanism for gathering the data (such as CD detector) can introduce noise.

Electronic transmission of image data can introduce noise.

To remove noise from image the different methods available are

Using Linear filtering

Using median filtering.

Using Adaptive filtering

A) Using Linear Filtering:

Filtering is a technique for modifying or enhancing an image.

Linear filtering is filtering method in which the value of an o/p pixel is a linear combination of the values of the pixels in the o/p pixel's neighborhood.

The above stated noise can be removed by using linear filtering. Certain filters such as averaging or Gaussian filters are appropriate for this purpose.

An average filter is useful for removing grain noise from a photograph, because each pixel gets set to the average of the pixels in its neighborhood, local variation caused by grain are reduced.

B) Using Median Filtering:

Median filtering is similar to using an average filter. However with median filtering the value of an o/p pixel is determined by the medium of the neighborhood pixels, rater than the mean.

The median is much less sensitive than the mean to extreme values (called outliers). Median filtering is therefore better able to remove the outliers without reducing the sharpness of the image.

Median filtering is a specific case of order-statistic filtering, also known as rank filtering.

C) Using adaptive filtering:

- Linear filters are used in many applications. A filter will be optimal only if it is designed with some knowledge of input data. If this information is not known adaptive filters are used. The adjustable parameters in the filter are assigned with values based on the estimated statistical characteristics of the signal.
- A wiener filter (a type of linear filter) applies to an image adaptively, tailoring itself to the local image variance. When the variance is large the filter perform little smoothing of image & where the variance is small, it perform more smoothing. This approach often produces better result than linear filtering; the adaptive filter is more selective than a comparable linear filter, preserving edges and other high frequency parts of an image.
- In addition, there are no design tasks, it handles all preliminary computation & implements the filter for an input image, however it does require more computation time than linear filtering.
- It works best when the noise is constant-power (“white”) additive noise, such as Gaussian noise. Adaptive filters find applications in adaptive noise cancelling, line enhancing, frequency tracking, channel equalizations, etc.