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Digital Communications: Introduction to Key Concepts and their relation to Acoustic Water Column Channels

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Outline- Key Concepts

- Features of Acoustic Water Column Channel
- Digital Communication
 - Communication without channel impairments
 - Channel Noise Impairments
 - Channel Attenuation and Fading Impairments
 - Channel Doppler Impairments
 - Channel Length Impairments
 - Channel Bandlimited Impairments
 - Modulation
- Summary
- Next Steps

Features of Acoustic Water Column Channel

- Bandwidth is approx. 0-100KHz
 - Wideband and baseband channel- not narrowband
- Very Long channel
 - For example over a 300m pipe could expect delay spread of 0.5-1 sec
 - Symbol of 0.1ms implies 5000-10000 channel symbol length
- Attenuation
 - Not well characterized but perhaps km propagation range possible

Features of Acoustic Water Column Channel

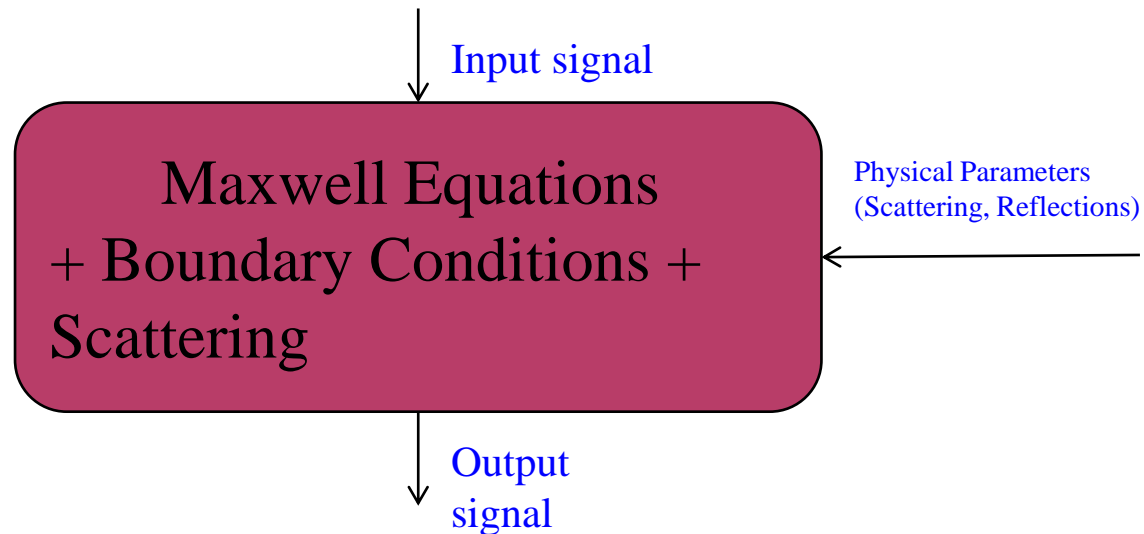
- Channel variations could be high
 - Carrier sync issues- relative frequency shift is very high compared to wireless?
 - Channel not pseudo-stationary: delay spread larger than coherence time?
 - c is low (1000m/s) so even small speed variations can provide large relative Doppler- $f' = (c+s)/c f$ where Rx approaching
 - For $s=1\text{m/s}$ relative Doppler frequency of 0.1% or 1Hz for 10KHz carrier
 - Moving reflector could be motor or valve
 - Wave speed could also change due to medium speed- water motion
- Noise and Interference
 - Not well characterized- interference at low frequencies such as vehicles and pumps

Lessons Learned from Wireless Communications



Wireless Communications Channels

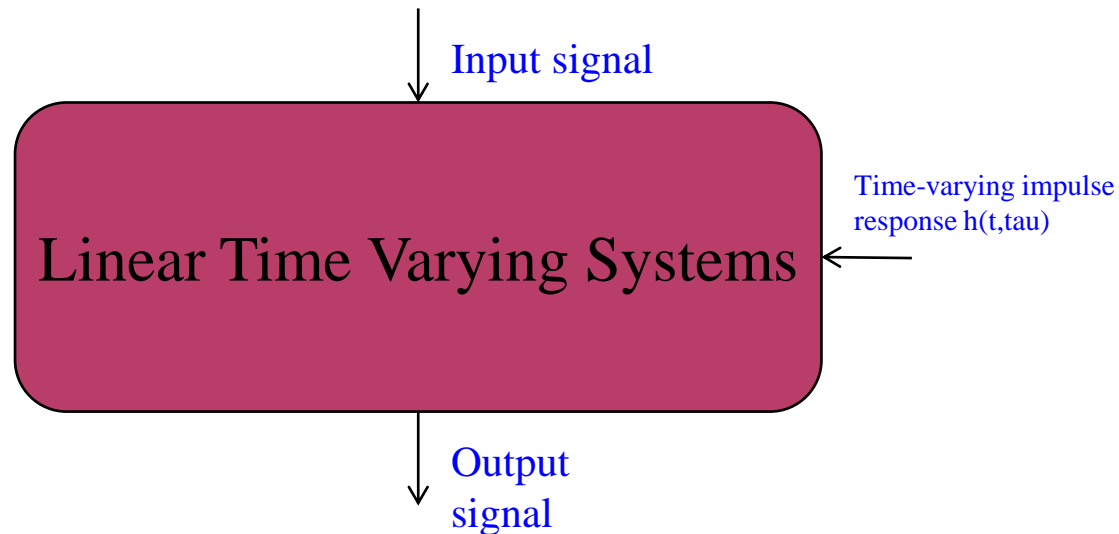
Propagation Physics Approach



Complicated (involve solving a bunch of PDE)c

Wireless Communication Channels

Model-based Approach



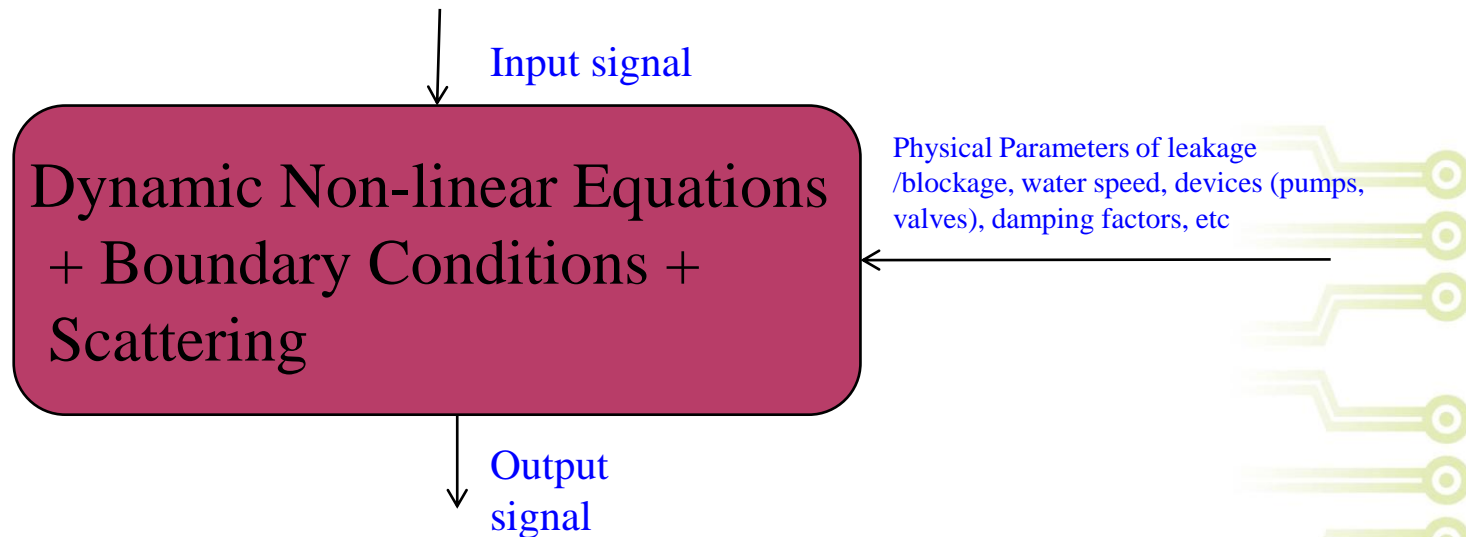
Since Maxwell equations are linear, the channel can be modeled as LTV system (characterized by model-based parameters \sim impulse response $h(t,\tau)$). Communications involves characterizations of these “model parameters” \rightarrow simple and intuitive. [Only characterize the “overall effects” of scattering / reflections / multipath]

[1] **Training phase:** Receiver characterizes $h(t,\tau)$ based on pilot channels / preambles ($h(t,\tau) \sim$ constant for coherence time [$\sim 5\text{ms} - 100\text{ms}$])

[2] **Communication Phase:** receiver detect symbols based on the estimated $h(t,\tau)$.

Acoustic Communications Channels

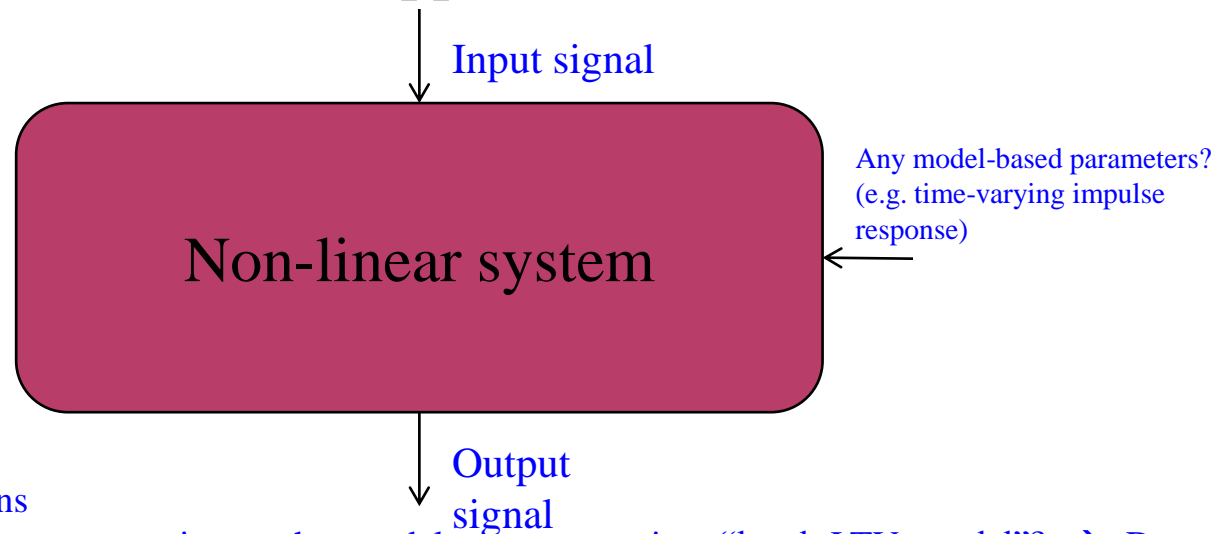
Propagation Physics Approach



Complicated (involve solving a bunch of non-linear PDEs)

Acoustic Communication Channels

Model-based Approach ??



For in-pipe communications

[Q1] Non-linearity: Can we approximate the model response using “local LTV model”? → Depends on the communication bit rate, range, ...etc.

[Q2] Uncertainty: Uncertainty on speed, device characteristics, damping, etc can be captured as random effects on the impulse response $h(t, \tau)$. → Automatically captured during channel training phase.

[Q3] Steady-state Analysis: For communications, what matters is the steady state response.

For inverse imaging

[Q1] Model-based vs Physics-based: Can we have a model relating the “blockage/ leakage” physical parameters with LTV impulse response? + Statistical inference on these blockage/leakage parameters based on the impulse response observations?

[Q2] Steady-state vs Transient Analysis: Any benefit of steady state vs transient analysis in imaging application?

[Q3] Uncertainty: Uncertainty on physical parameters (e.g. speed, device characteristics, damping) can be captured by statistical inference.

Digital Communications: Communication without Channel Impairments

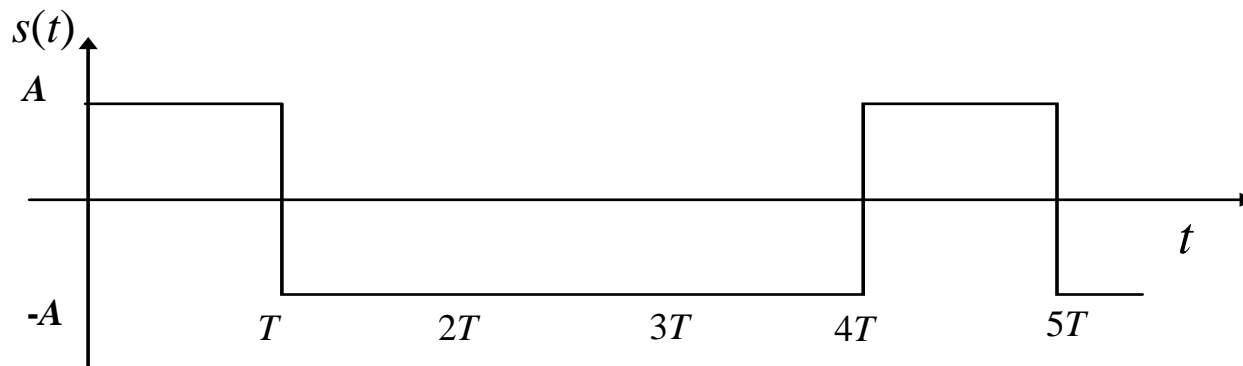


Digital Binary Signals

- Communication is the process of transmitting information from place to another
- All information or messages or signals, no matter how complicated, can be converted into digital binary- 0 and 1's
- There can be some loss of signal accuracy but this can be made arbitrarily small if need be
- Digital binary signals are great because they can be processed cheaply and conveniently by computers, DSP, embedded systems etc
- They can also be stored cheaply and easily in memory
- We can represent binary signals by electric signals by mapping 1's to say A volts and 0's as $-A$ volts for example

Digital Binary Communication

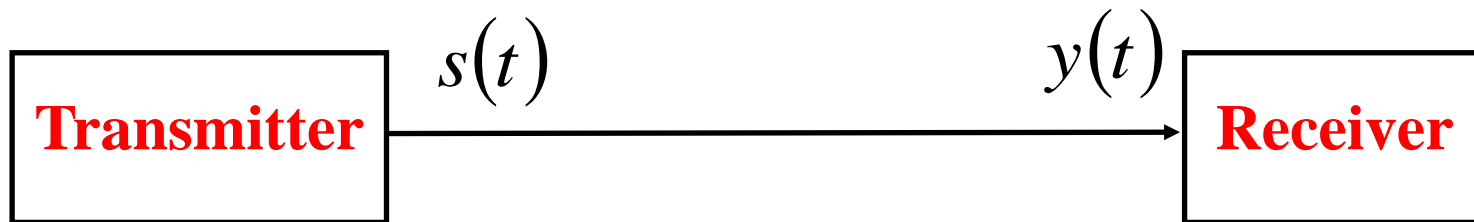
- We can transmit binary over wires easily by mapping 1's to say A volts and 0's as $-A$ volts
- The sequence 1,0,0,0,1,0 maps to the electric signal



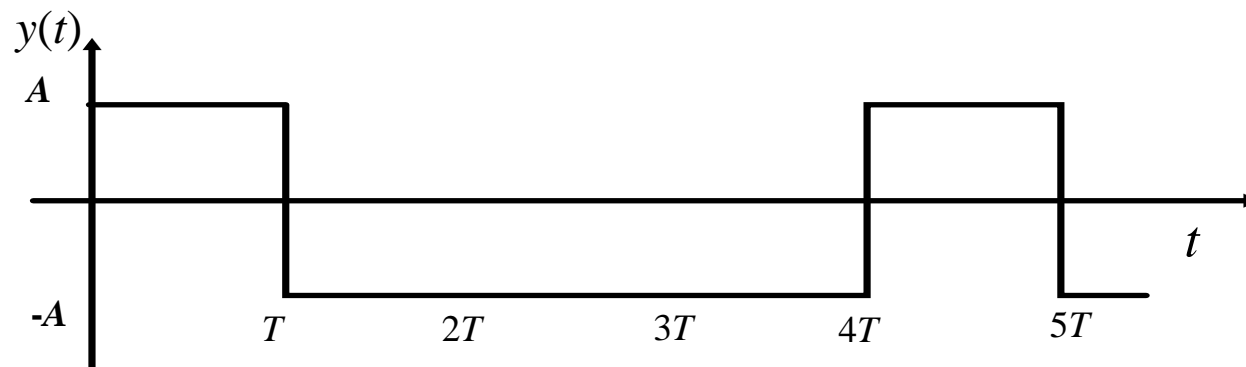
- We refer to this as the signal $s(t)$
- The bit duration is T and $R = 1/T$ is the bit rate in bits/sec

Communication

- We can transmit these over wires (or simple channel) easily



- We refer to $s(t)$ as the transmitted signal and $y(t)$ as the received signal
- Since $y(t)=s(t)$ the transmitter can easily detect the A and $-A$ by sampling every T seconds and then map back to 0 and 1

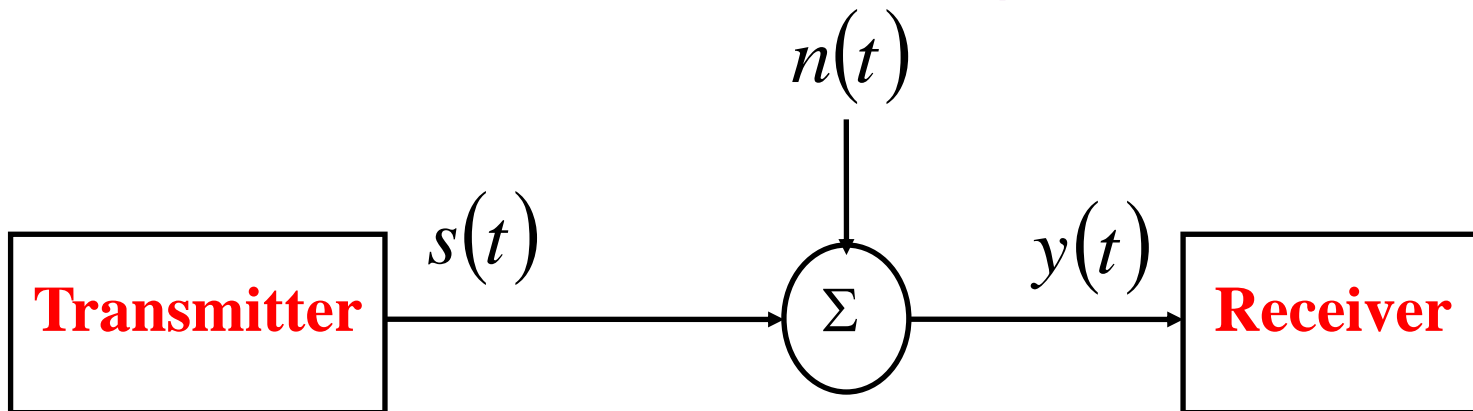


Digital Communications: Channel Noise Impairments

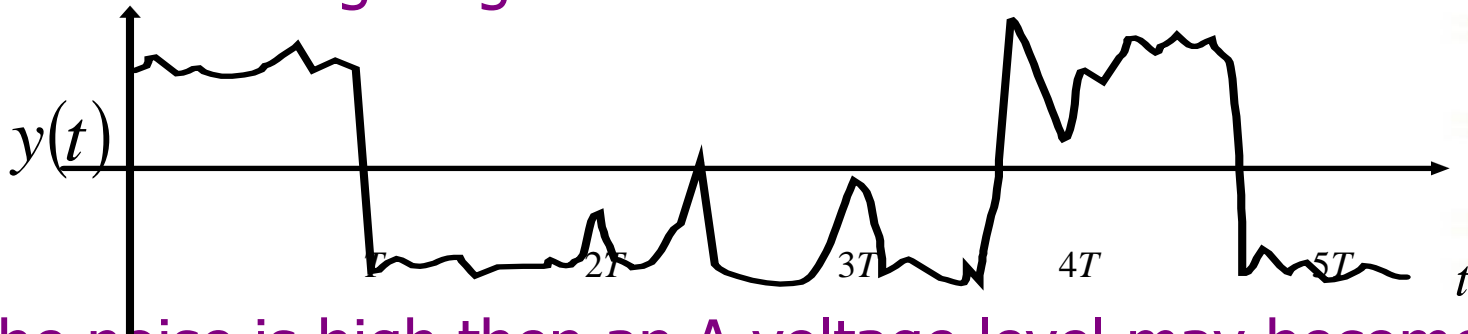


Why is Communication Difficult?

- Realistic channels add noise and a possible model is



- The received signal gets distorted



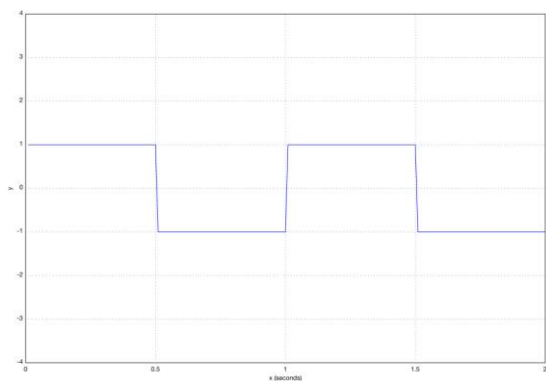
- If the noise is high then an A voltage level may become < 0 at the receiver creating errors

Digital Signal Quality

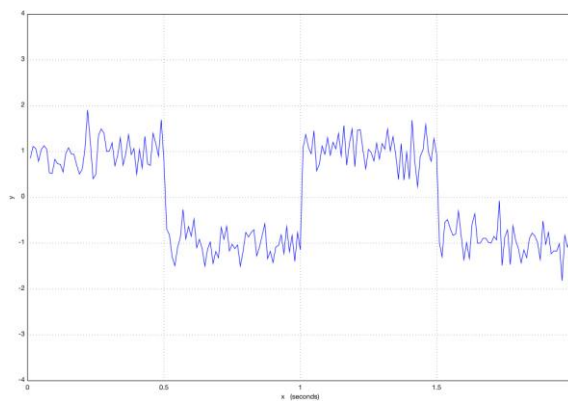
- The quality of the received signal can then be measured by how many bit errors we get and we refer to this as the Bit Error Rate or BER
- We want the BER to be low as possible and for voice transmission we want it to be better than 0.001
- This means that for each thousand bits sent we can expect on average one bit error
- For data applications we need to make the BER even smaller and handshake methods developed to reduce the errors to zero
- BER is closely related to the Signal-Noise-Ratio (SNR) of the received signal $y(t)$

SNR

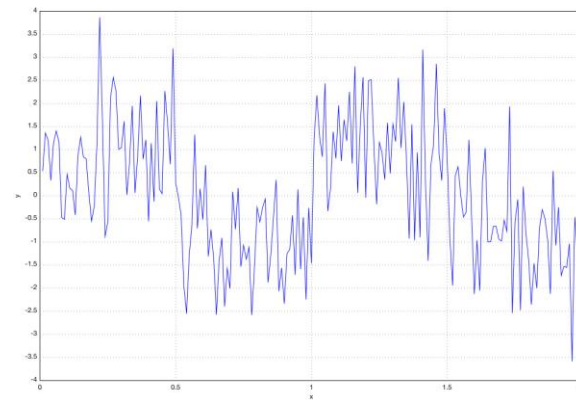
- Examples for SNR of 0 and 10dB are shown below



Signal with No Noise



10dB SNR



0dB SNR

- Generally you need an SNR of better than 10dB to have reliable communications

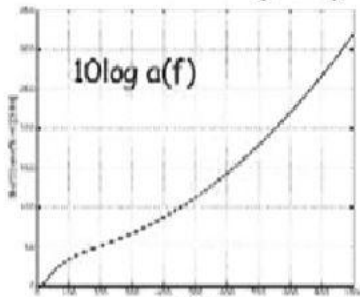
Where is the Noise coming from?

- The electronic noise is from the background thermal activity and it is approximately Gaussian distributed with zero mean
- We therefore refer to these channels as AWGN- Added White Gaussian Noise
- White refers to the idea that it is of equal power density at all frequencies across the radio spectrum
- Received noise power is found from $P_n = KTB$ where B is the receiver bandwidth, K is Boltzmann's constant and T is temp
- For a bandwidth of 1MHz P_n is 4×10^{-15} Watts
- For reliable communication the desired signal should have a power x10 of that or 4×10^{-14} Watts
- This is a very very small power!! However your mobile today is able to communicate at these levels!!

Acoustic Noise and Interference

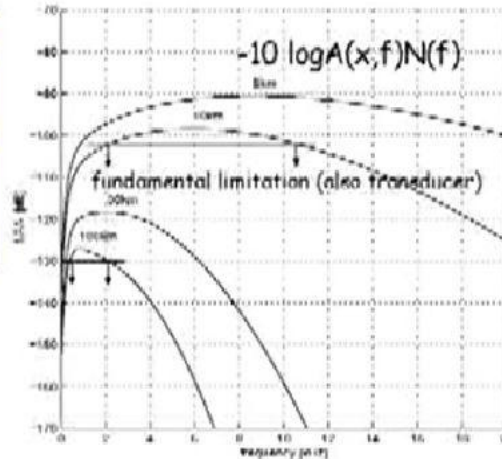
Underwater acoustic channel: attenuation and noise

$A(x, f) \sim x^k a^x(f)$ spreading + absorption
 (k:1-2)

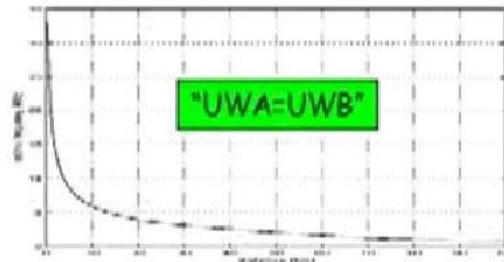
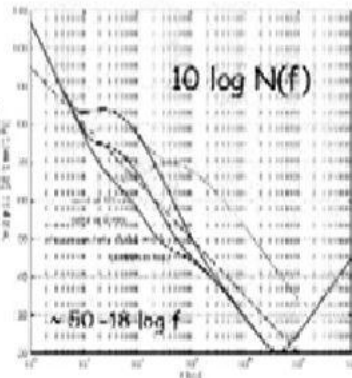


$10 \log a(f) = 0.11 f^2 / (1 + f^2) + 44 f^2 / (4100 + f^2) + 0.000275 f^2 + 0.003$
 dB/km, for f [kHz]

$SNR = \frac{P_x}{P_{noise}} \sim \frac{P/A(x, f)}{N(f)\Delta f} \sim \frac{1}{A(x, f)N(f)}$



noise =
 turbulence +
 shipping +
 surface +
 thermal +
 other
 ↓
 site specific:
 man-made
 biological
 ice, rain
 seismic



- Prof. Milica Stojanovic,
<https://www.youtube.com/watch?v=k8ZqDnfNx4I&index=2&list=WL>
 , slide at 16min/51.52min

Acoustic Noise and Interference

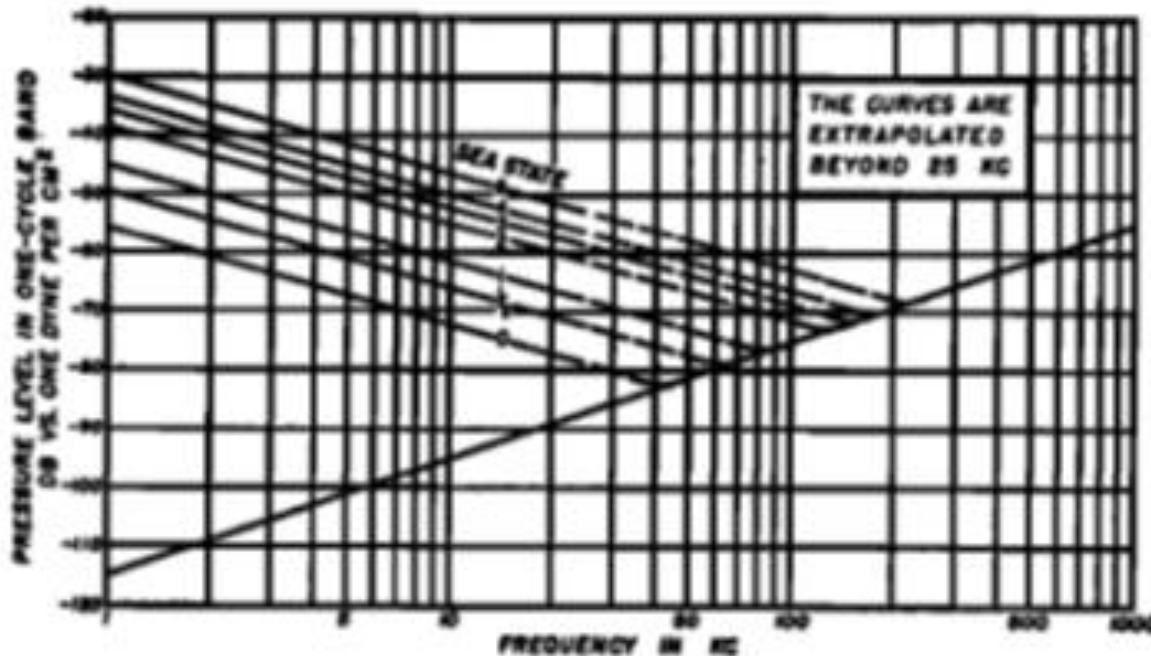


FIG. 1. Comparison of thermal-noise spectrum and ambient-noise spectra.

- RH Mellen, The Thermal-Noise Limit in The Detection of Underwater Acoustic Signals, JASA, Vol 24, No 5, P478-480, September 1952

Noise provides a Limit

- Noise power sets the lower limit to the power we need to transmit a signal
- The minimum signal power needs to be about x10 the noise power
- Determines range of communication, battery life, capacity of power transmitter etc
- We therefore need to make tradeoffs
- To make these tradeoffs we need to know the BER
- We spend much effort in determining BER verses for a new receiver structure as it determines some of the tradeoffs

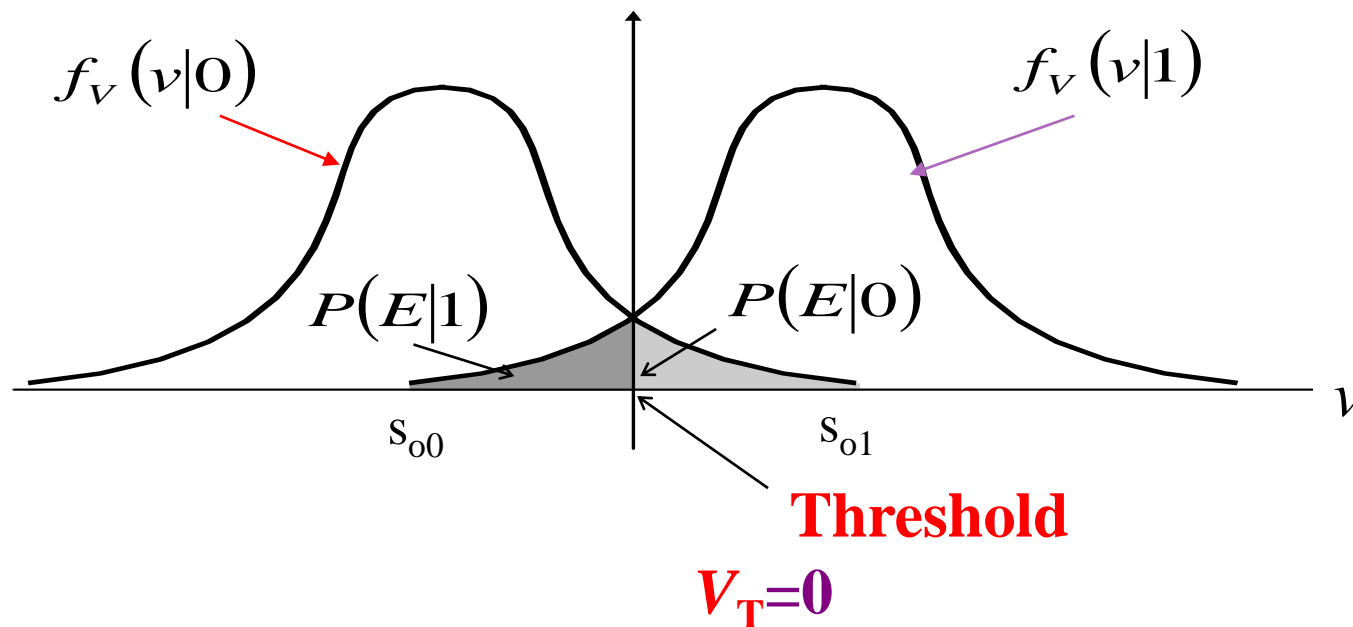
Quick Example

$$\mathbf{V} = \begin{cases} s_{o1} + N & \text{if "1" is sent} \\ s_{o0} + N & \text{if "0" is sent} \end{cases}$$

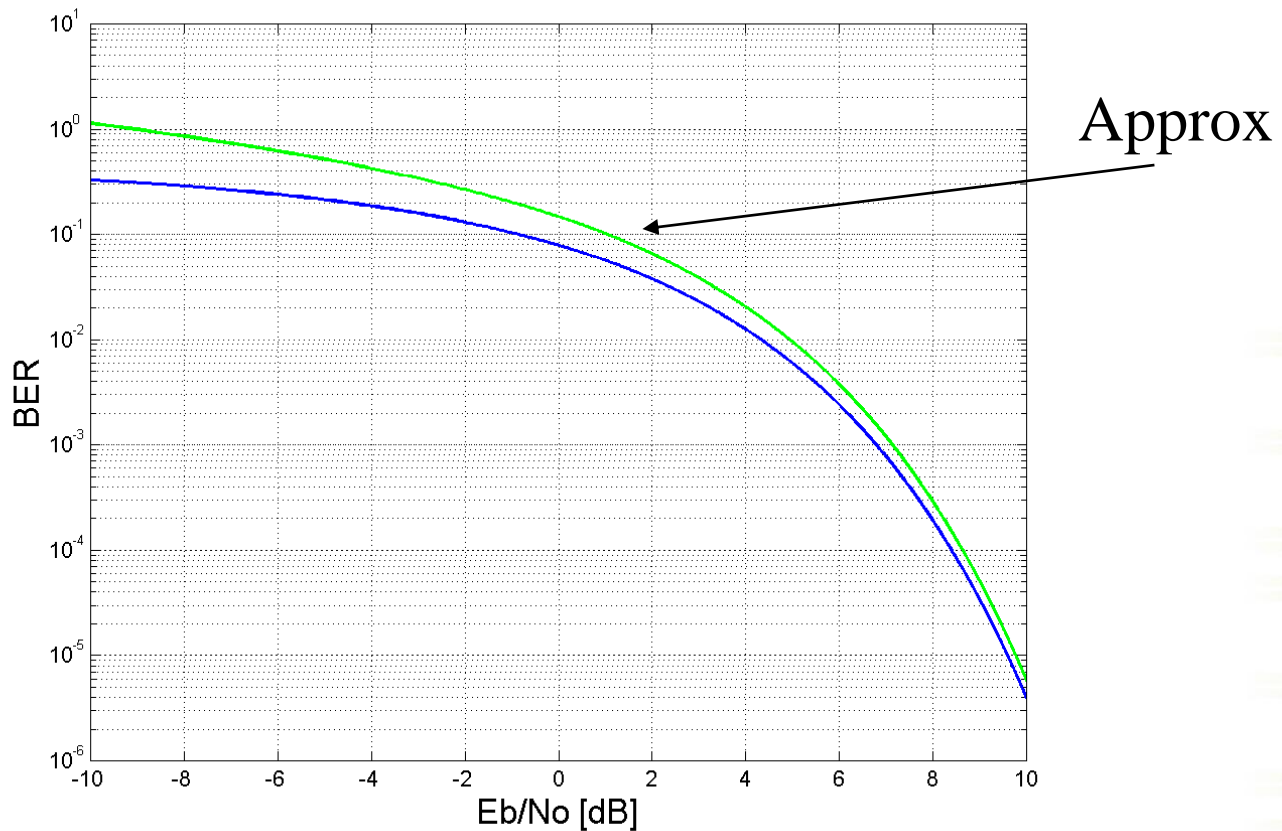
AWGN (Zero Mean)

$$\left. \begin{array}{l} E[V|1] = s_{o1} \\ \text{Var}[V|1] = \sigma^2 \end{array} \right\} \Rightarrow f_V(v|1) \sim N(s_{o1}, \sigma^2)$$

$$\left. \begin{array}{l} E[V|0] = s_{o0} \\ \text{Var}[V|0] = \sigma^2 \end{array} \right\} \Rightarrow f_V(v|0) \sim N(s_{o0}, \sigma^2)$$



A graph of P_e for **baseband signaling** is



where

$$P_e = Q\left[\sqrt{2E_b / N_0}\right] = \frac{1}{2} \operatorname{erfc}\left[\sqrt{E_b / N_0}\right]$$

Example:

A baseband digital Tx system sends $\pm A$ valued rectangular pulses through a channel at a rate of 1Mbps with amplitude 1V when the noise single-sided PSD is $2 \cdot 10^{-7}$ W/Hz.

Answer: $Q\left(\sqrt{2E_b / N_0}\right) = Q\left(\sqrt{2A^2T / N_0}\right)$

$$T = 1/1000000 = 10^{-6}$$

$$\Rightarrow Q\left(\sqrt{2 \times 10^{-6} / (2 \times 10^{-7})}\right) = Q(\sqrt{10}) = Q(3.16)$$

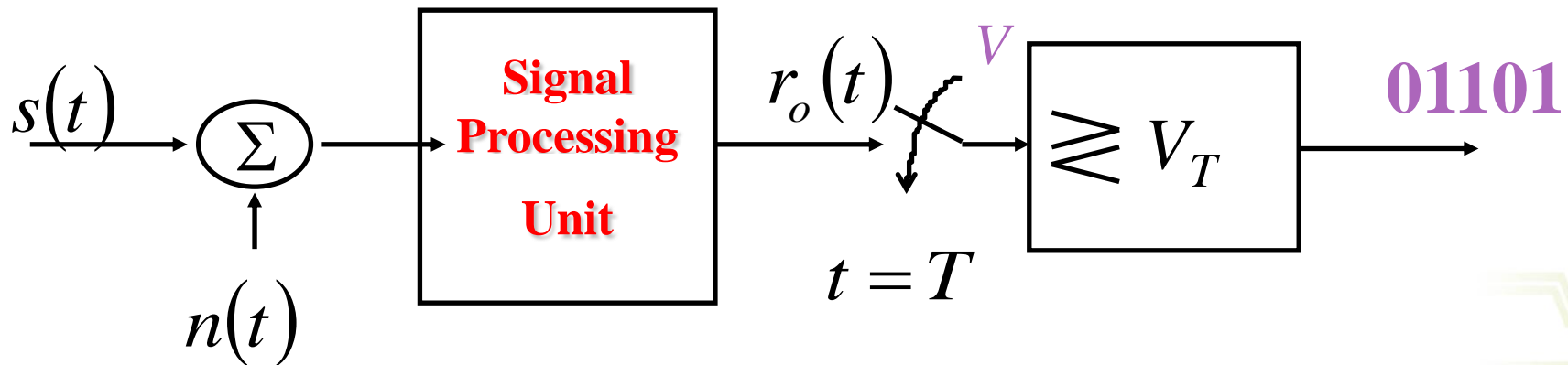
$$Q(u) \approx \frac{e^{-u^2/2}}{u\sqrt{2\pi}}$$

$$Q(x) \approx \frac{1}{2}e^{-\frac{x^2}{2}}, \quad x > 3$$

$$Q(3.16) \approx 0.00085$$

Optimum Receiver Structure

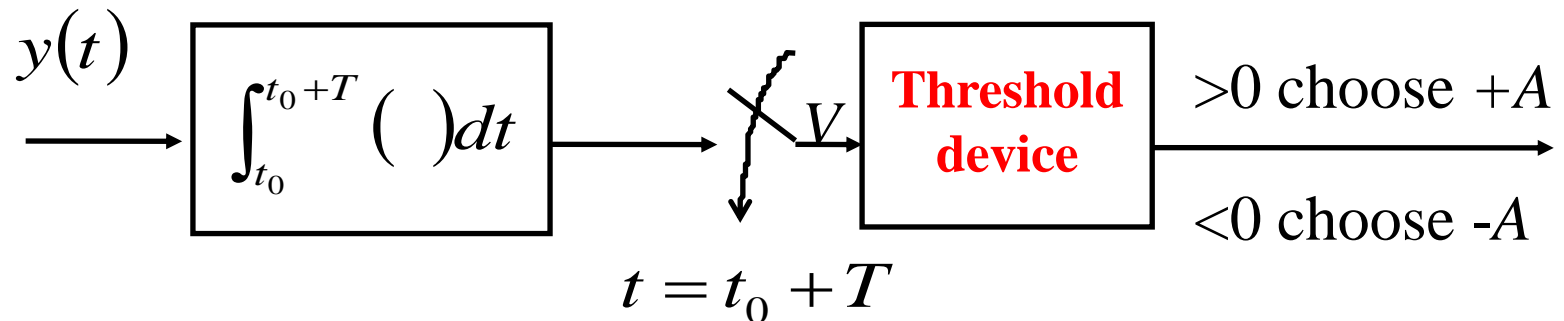
- Develop receiver which minimizes BER- Optimum



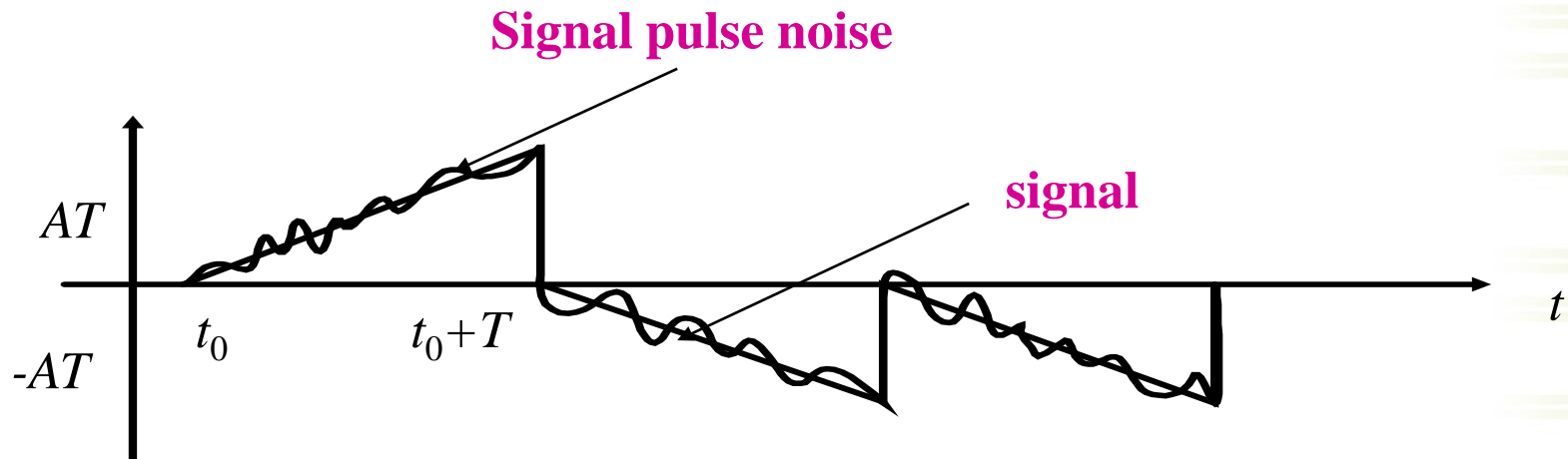
- V_T is a **threshold**.
 - If received signal (sampled every t_0) $> V_T$. Then, we decide that “1” was transmitted. Otherwise, we decide that “0” is transmitted.

What is the optimal detector to be used for the Signal Processing Unit?

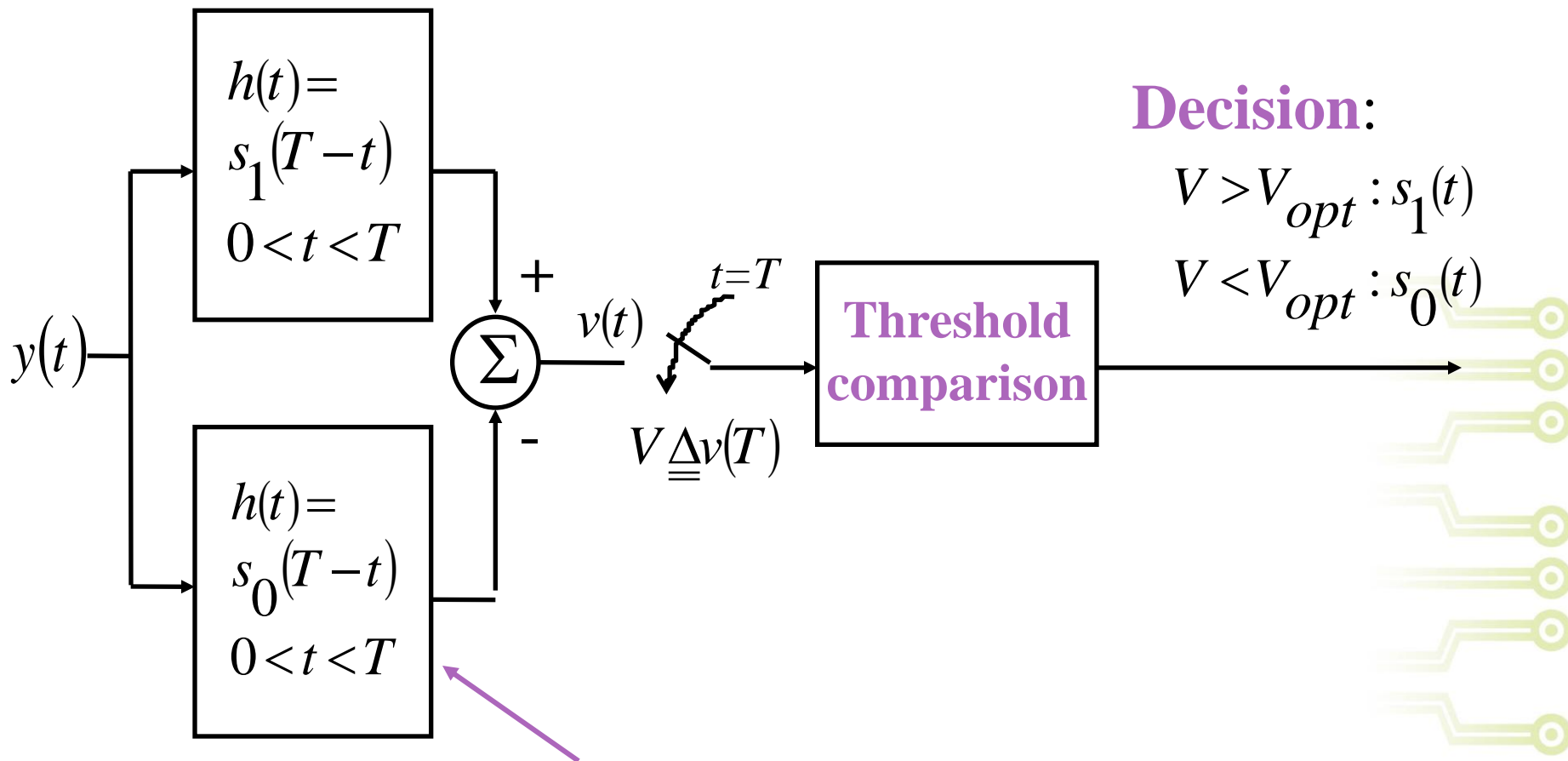
A possible receiver structure for detecting the digital transmitted signals is shown below



- The **integrator** averages out the noise received so that the output waveform will look like



Optimum (Matched filter) receiver for binary signaling in white Gaussian noise



2 Matched Filters (each matched to $s_1(t)$ and $s_0(t)$)

Matched Filter

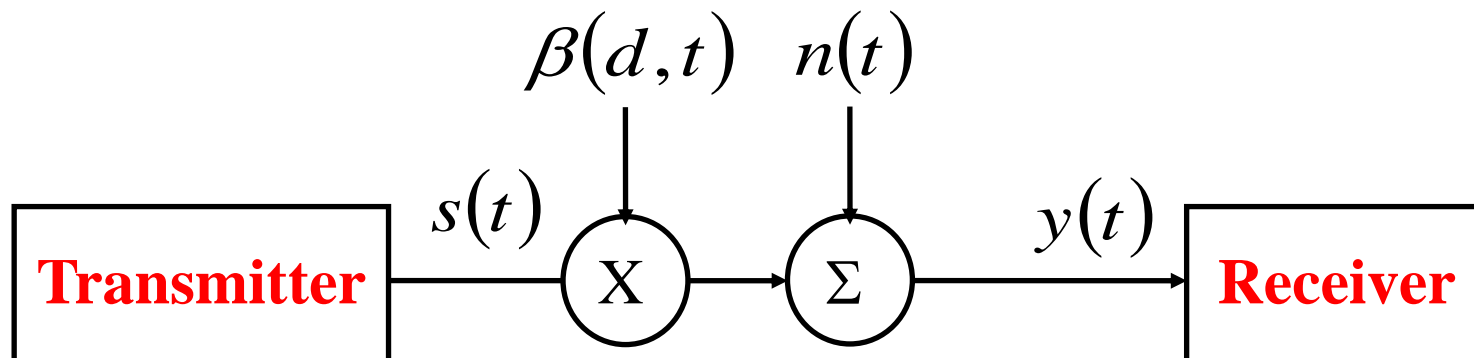
- The optimum filter $h(t)$ for detecting a certain signal $s(t)$ in AWGN is $h(t)=s(T-t)$
- In the frequency domain it is easier to understand
- Magnitude of matched filter is same as signal $|H(f)| = |S(f)|$
- Phase is conjugated $\angle H(f)=\angle S(f)^*$
- This is why it is called a matched filter
- At frequencies where signal is large filter is large
- The conjugate phase of signal in filter allows the received signal to add up coherently at each frequency maximizing its power

Digital Communications: Channel Attenuation and Fading Impairments



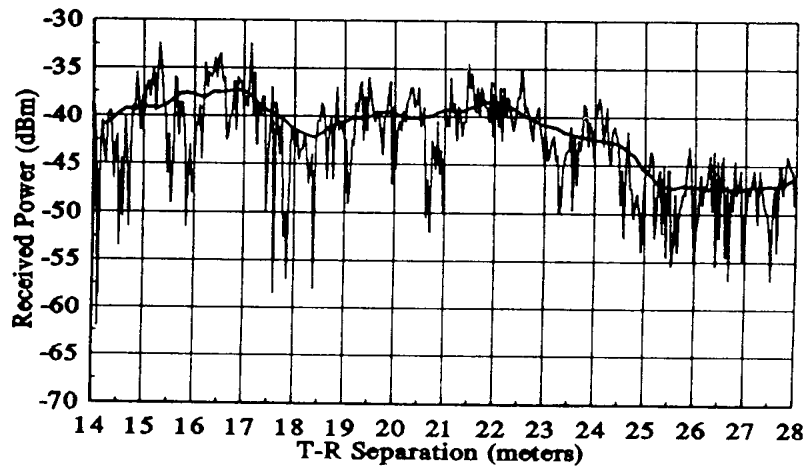
More Channel Effects-Attenuation and Fading

- The Channels can also have attenuation
- In wireless this attenuation is usually split into 3 effects over different scales
 - Path Loss Model } **Large-scale propagation**
 - Shadowing
 - Multipath Fading → **Small-scale fading**
- Can be modelled as a multiplicative attenuation that can be a function of time and distance between Tx and Rx

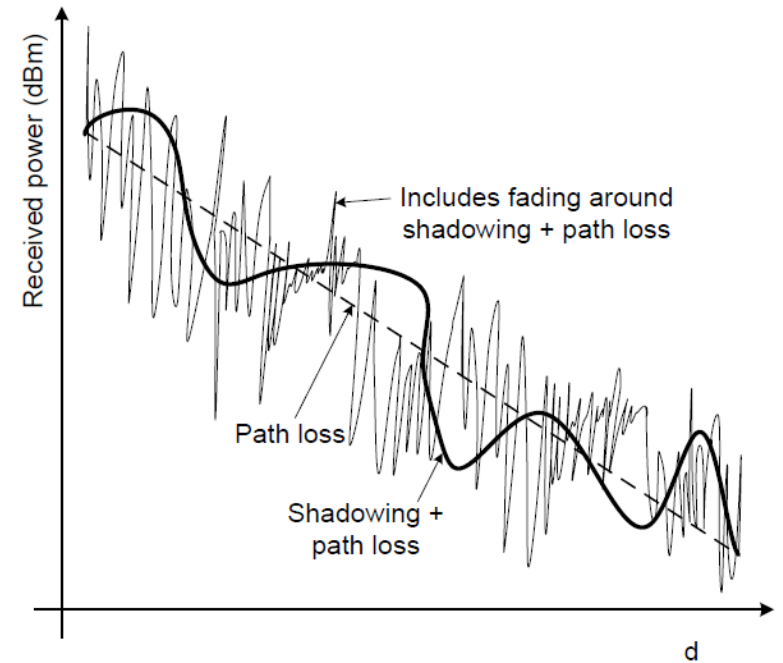


3-level Model

Over short distance



Over long distance



Digital Communications: Channel Doppler- Channel Variations



Doppler

- The speed at which $\beta(t,d)$ changes defines the Doppler shift of the received signals
- It is caused by things moving such as valves and motors or by moving the receiver or transmitter itself which changes d with time or by the relative speed of the medium in acoustics
- If Rx is moving towards Tx at speed s then the frequency will appear to increase by $f \times s/c$ where c is speed of acoustic wave and f Tx frequency
- Since acoustic speed c is very slow compared to that of light, Doppler could be a much bigger problem in the acoustic case than RF case
- If c is 1000m/s and movement is 1m/s then Doppler shift can be up to 0.1% which is very big compared to wireless

Doppler and Coherence Time

- If we denote max Doppler frequency shift as f_d then we can approximately think of the channel as changing over a time scale $1/f_d$ which is known as the coherence time
- The reason for this is that relative to the carrier, the Doppler shift would have caused a full 360 phase shift over the time period of the coherence time
- Therefore possible constructive and destructive combining and therefore channel variation occur
- If the Doppler is 1Hz for a carrier of 10KHz then we can expect the channel to change significantly over a period of 1 second
- For acoustics the speed of the water flow will also affect the frequency shift but as long as its time variation is low may not directly relate to channel variation

Digital Communications: Channel Length Impairments- Delay Spread and ISI



More Channel effects- Inter-Symbol Interference (ISI)

- The channel model is further generalized to include multiple paths each with a different delay
- Delay spread caused by multipath fading

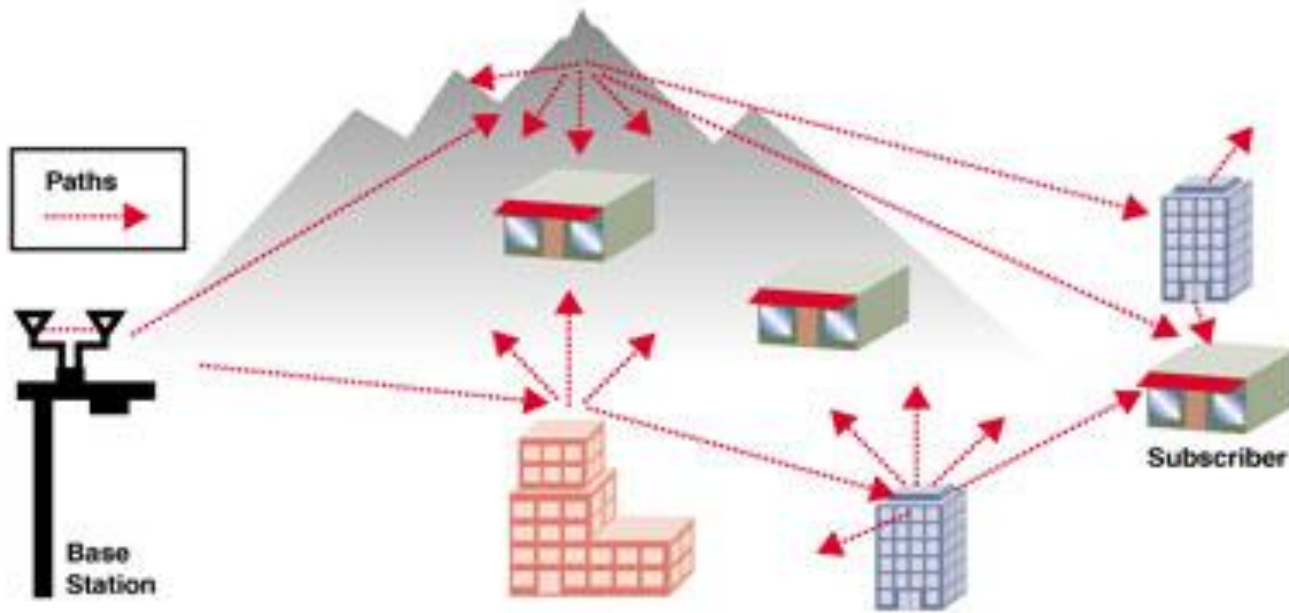
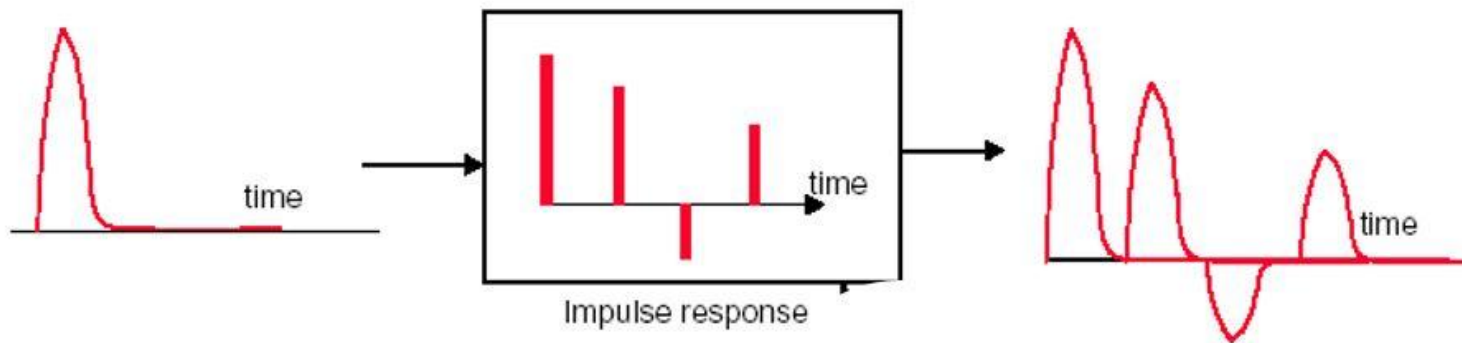


Fig. 1 Typical multipath.

Time Domain Description of Multipath

Time domain: Impulse response

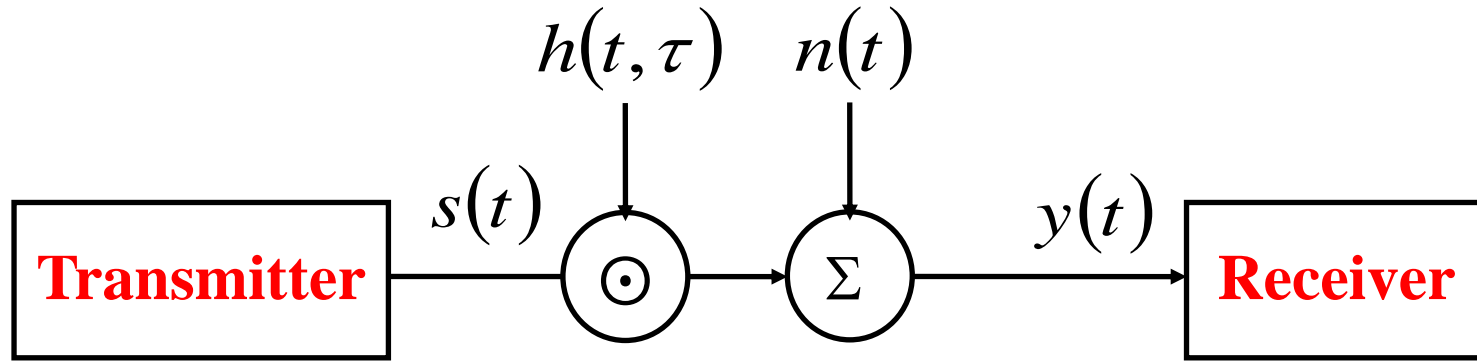


$$h(t, \tau) = \sum_{p=1}^P \beta_p(t) \delta(t - \tau_p)$$

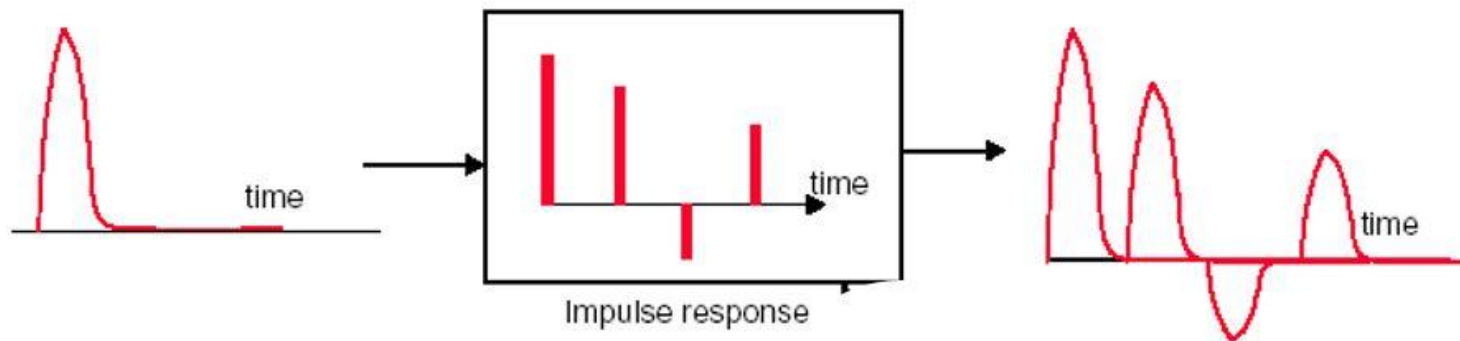
- Note that the multiple paths each have a delay τ_p and vary with time t
- The basic model previously only has one path

ISI Channel Model

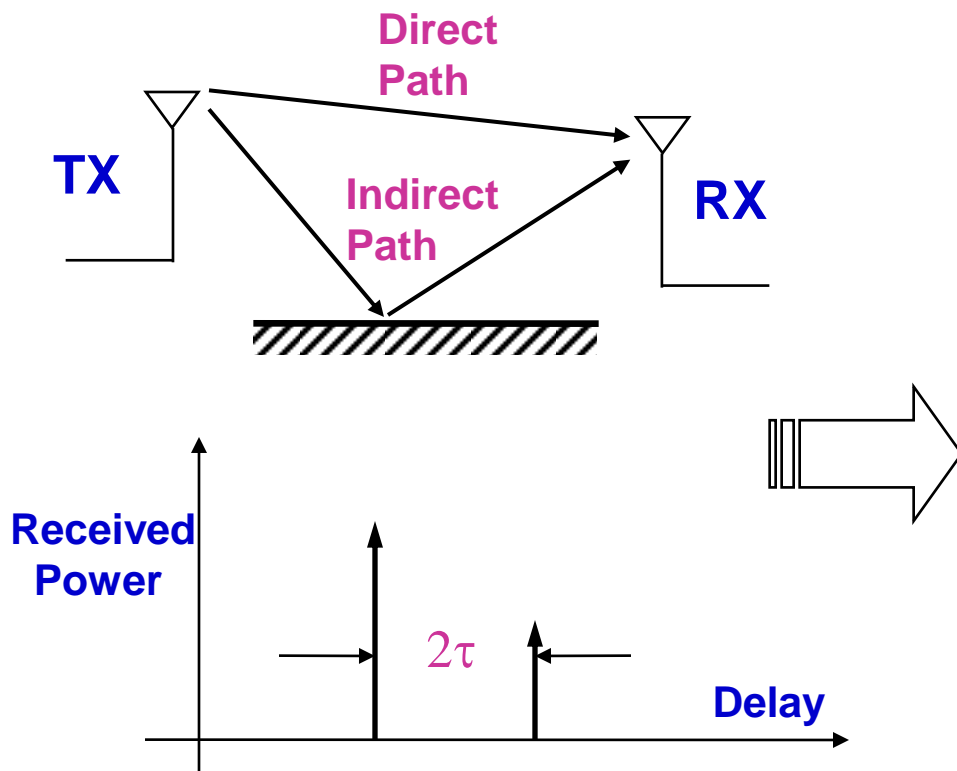
- The multipath fading changes the single path multiplicative fading to a convolution with the channel impulse response



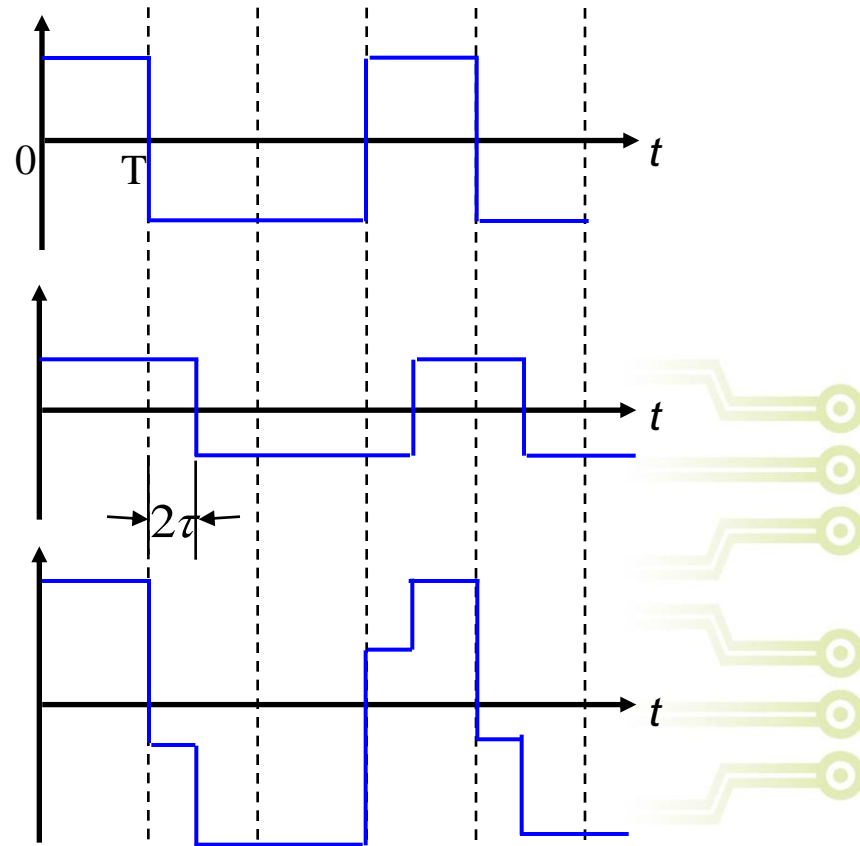
Time domain: Impulse response



Effect of ISI



Two-ray gain profile
 τ is rms delay spread



τ/T small \rightarrow negligible ISI
 τ/T large \rightarrow severe ISI

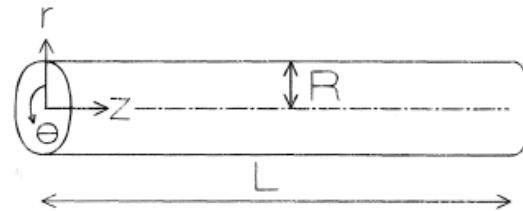
ISI Effects

- We refer to the maximum delay of the channel as its channel length τ or delay spread
- If $\tau/T > 1$ then there will be significant ISI causing errors
- If channel length is long we reduce the bit rate increasing the bit period T
- Like talking in a room with lots of echo- we slow down
- Alternatively we can use signal processing approaches to overcome the ISI
- One approach is known as equalization
- OFDM is used in 4G is OFDM

Acoustic Communications in Pipes

- Pipes become waveguides

- For canonical circular pipe we arrive at Bessel function solutions in radial direction and trigonometric functions in angle and we get propagation in modes m,n



- One difference from electromagnetic case is that the boundary condition $\partial\phi/\partial r=0$ gives a $0,0$ mode which is a planar wave with no cutoff frequency
- Used in most water pipes today for imaging as it can propagate at low frequency
- If HFW techniques used then higher modes will occur

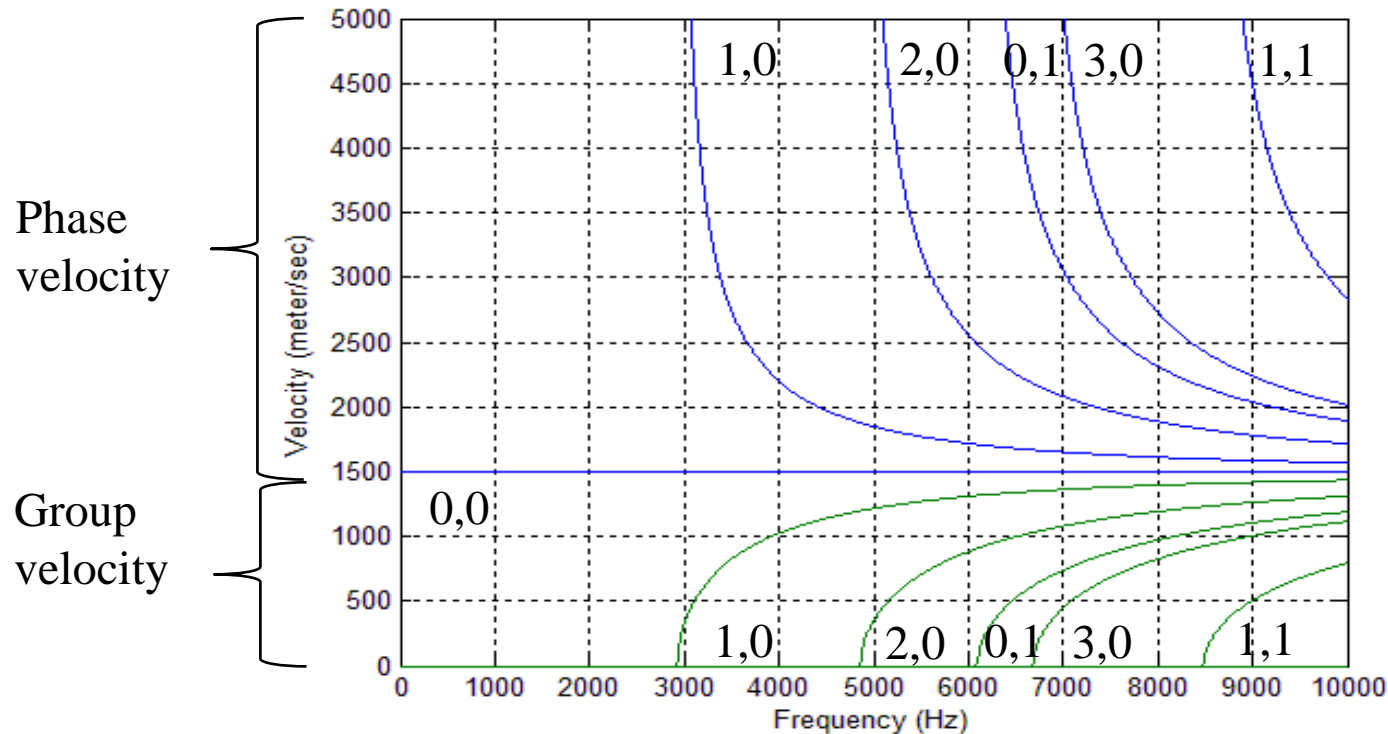
Propagation Modes

- In order to satisfy the boundary conditions ω and k become related by the dispersion relation $\omega = \omega(k)$
- For circular pipes can be found from $k_z^2 = k_0^2 - k_{nj}^2$ where k_0 is the wavenumber ω/c and $k_{nj} = z_{nj}/R$ and z_{nj} are the zeros of the Bessel functions of order n .

- Dispersion relation becomes $\omega = c \sqrt{k_z^2 - k_{nj}^2}$
- Derivative of dispersion relation is then the group velocity

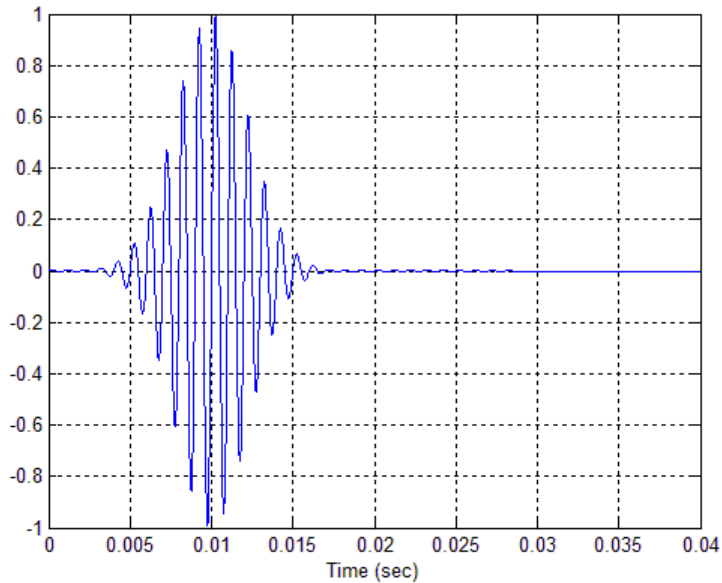
$$\frac{d\omega}{dk}$$

Mode analysis

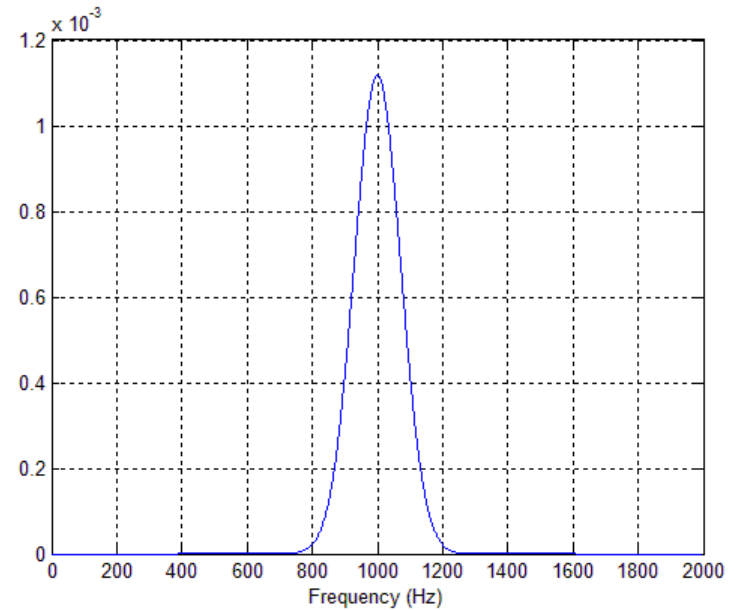


For different mode, the group velocity will also be different. For example the (0,0) mode have the highest velocity of 1500m/s. As the frequency increased and more modes are excited, the speed of higher order mode is getting slower.

Input signal

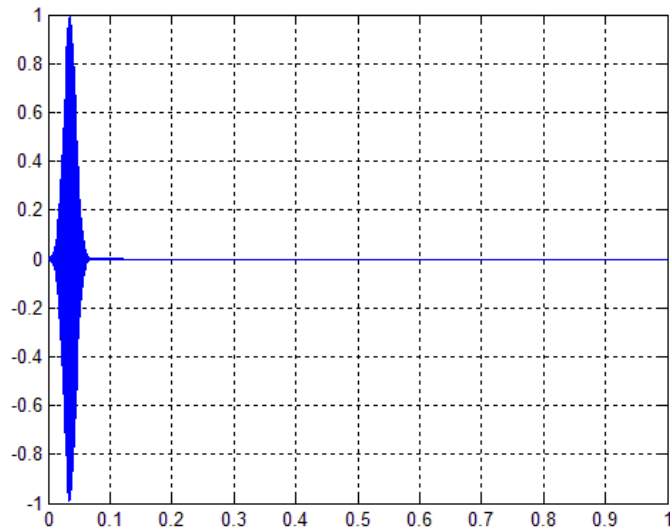


Input signal in time domain

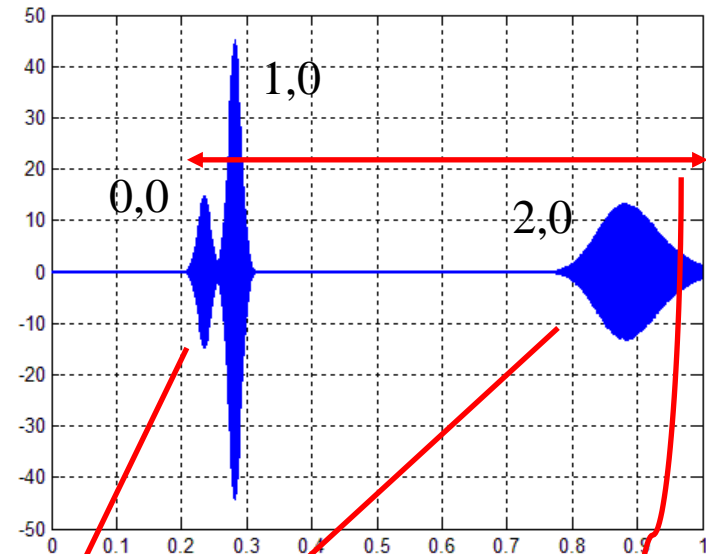


Input signal in frequency domain

Mode analysis



Input signal



Output signal

The frequency of input signal is 5KHz. The distance between transmitter and receiver is 300m. The radius of the pipe is 0.15m. In this case three modes (0,0) (1,0) (2,0) are excited. The different group velocity cause the time spread.

First two modes are traveling approximately at c and therefore arrive in about 0.35 secs

Third mode arrives much later around 0.9 seconds giving a channel length of around 0.8 secs

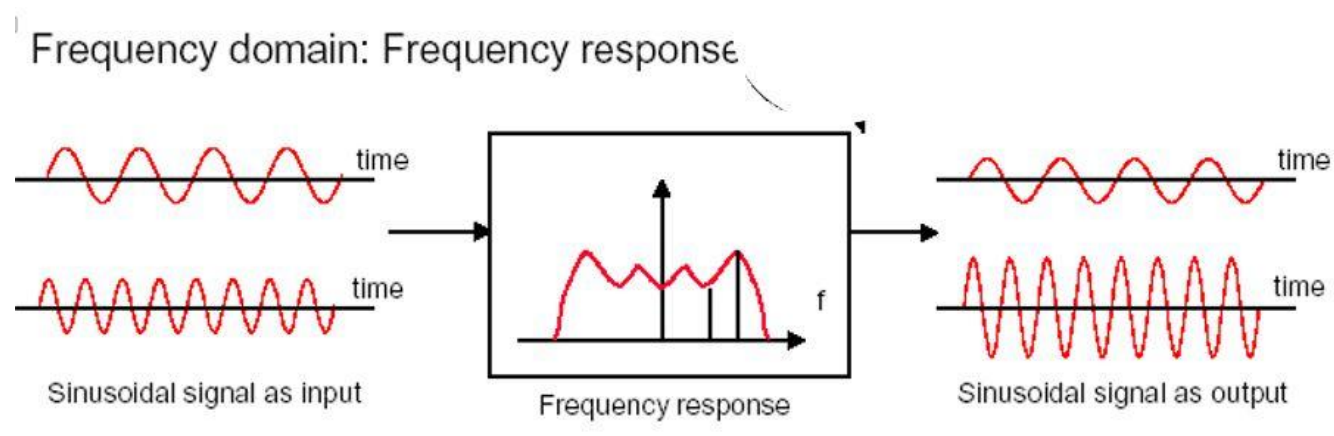
Delay Spread

- Significant delay spread occurs because of dispersion
- Around a delay spread of 0.8 seconds
- As pipe diameter increases more modes can propagate causing increased delay spread
- As frequency increases more modes can propagate causing increased delay spread
- However pipes also have bends, valves and other intrusions causing reflections increasing delay spread even further



Frequency Domain Description of Multipath

- Take Fourier Transform wrt to τ to get a visualization in the frequency domain



- ISI channel is the same as frequency selective fading in frequency domain
- The bandwidth over which the channel is flat is known as the coherence bandwidth

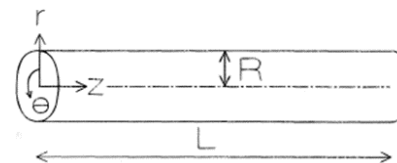
Coherence Bandwidth

- The coherence bandwidth can be thought of as the bandwidth over which the channel does not change and denoted B_c
- $1/B_c$ is approximately the channel length
- For the 300m pipe channel the channel length is approx 1 sec so the coherence bandwidth is 1Hz which is very low
- This means we could send a digital signal with a very low bit rate of say 0.5 bits per second and experience no ISI
- Even though the bit rate is low in principle we could send such a signal over each 1Hz band in our spectrum
- The problem is however is that the channel may change every second if the carrier is 10KHz

Model Assumptions

- Background

- Similar to principles of electromagnetics but with differences!
- Particle velocity v and pressure p characterize the acoustic waves
- Should not be confused with wave speed of propagation or ambient pressure of water
- With simplifying assumptions we end up with the standard Helmholtz wave equation $\nabla^2 p + k^2 p = 0$
- Can do the same for v by introducing velocity potential $v = \nabla \phi$
- Provides $\nabla^2 \phi + k^2 \phi = 0$ and has advantage that boundary condition at pipe boundary can be written easily

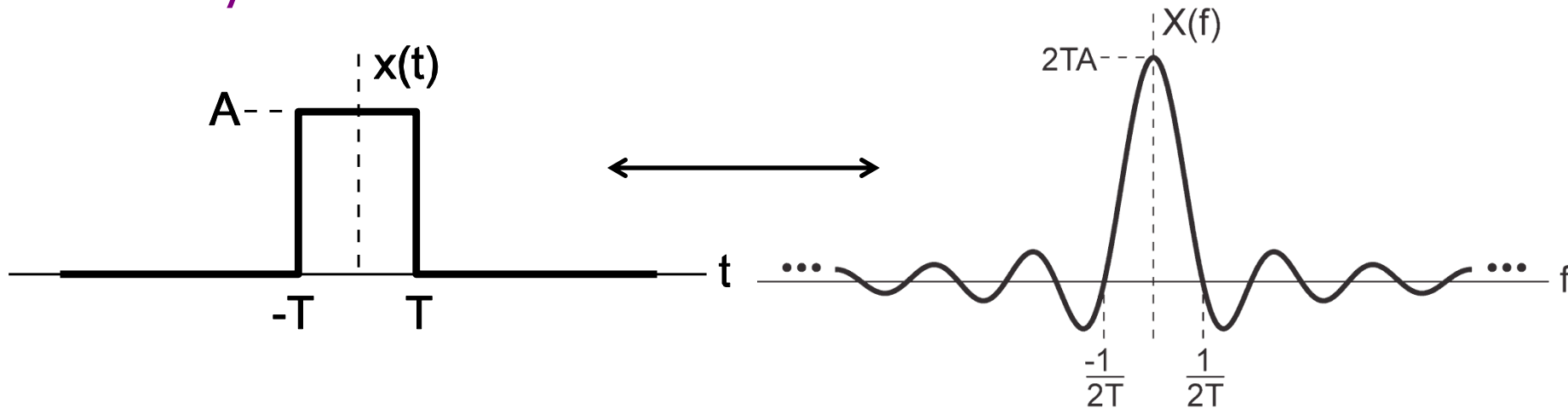


Digital Communications: Channel Bandlimited Impairments



Bandlimited Channel

- Channels can often only be used over a finite bandwidth
- The shape of the signals we transmit therefore needs to be carefully selected



Rectangular pulse

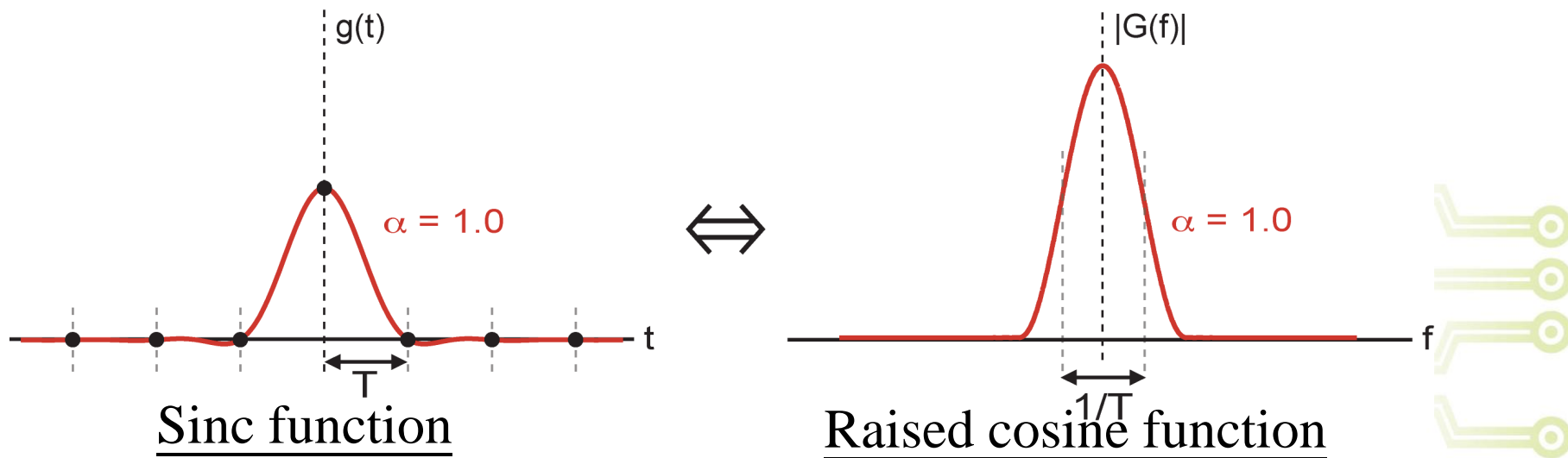
Sinc freq spectrum

(**Pros:** *no* interference during the sampling time of other pulses)

(**Cons:** *unbounded* frequency response renders it *unsuitable* for band-limited transmissions)

Raised cosine pulse

Ans. The raised cosine pulse, used in a wide variety of modern data transmission systems



- The raised cosine pulse takes on the shape of a sinc pulse in the time domain, and the shape of a raised cosine in the frequency domain.

Raised Cosine Filter Frequency Spectrum

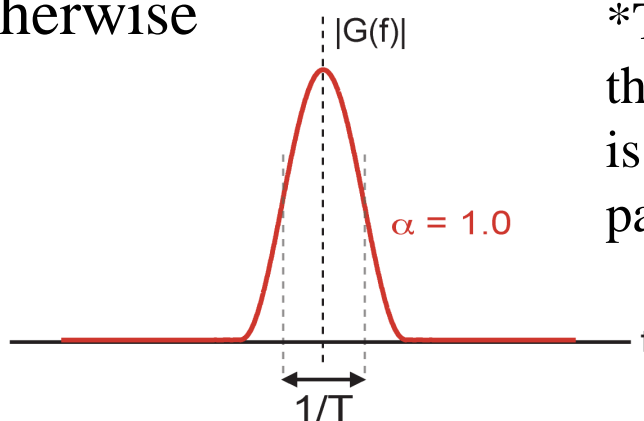
The frequency spectrum $G(f)$ is given by three piece-wise continuous functions

$$(1) G(f) = T, \quad |f| \leq (1-\alpha)/2T \quad (0 \leq \alpha \leq 1)$$

$$(2) G(f) = (T/2) [1 + \cos((\pi t/\alpha)(|f| - (1-\alpha)/2T))] \\ \text{for } (1-\alpha)/2T \leq |f| \leq (1+\alpha)/2T$$

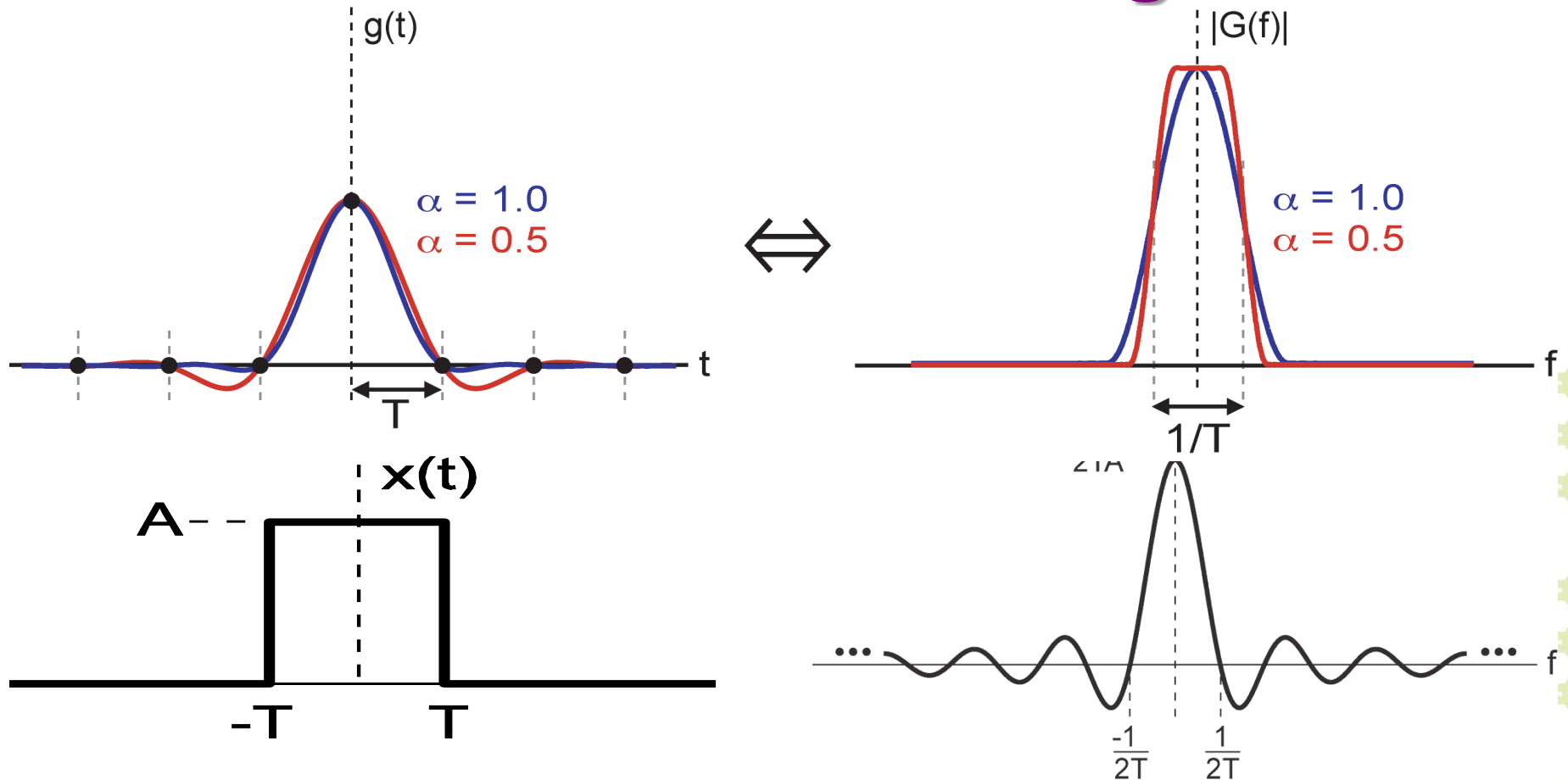
$$(3) G(f) = 0$$

otherwise



*The precise shape of the raised cosine spectrum is determined by the parameter α , $0 \leq \alpha \leq 1$

Raised Cosine Pulse vs. Rectangular Pulse

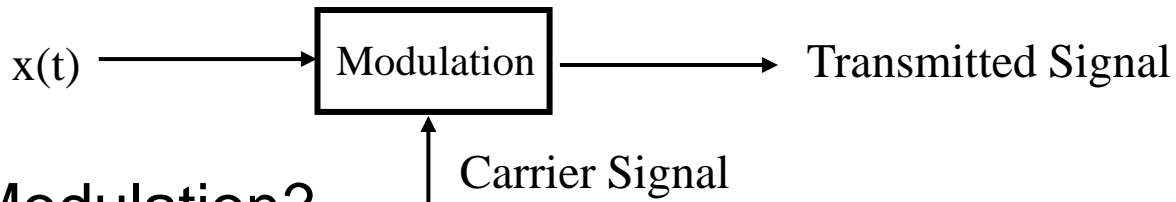


Raised cosine filter achieves low bandwidth and zero ISI

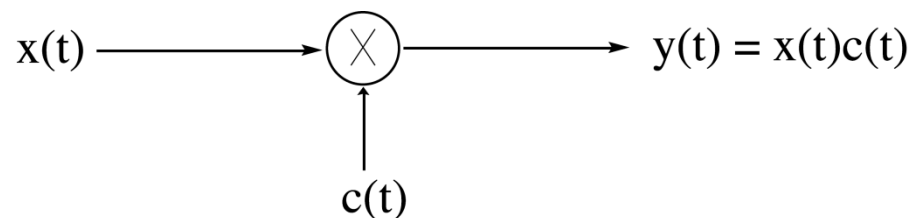
Digital Communications: Modulation or Frequency Translation



The Concept of Modulation



- Why Modulation?
 - Easier to transmit electromagnetic waves at higher frequencies
 - Transmitting multiple signals through the same medium using different carriers- multiplexing
 - Fitting signals to “channels” with limited passbands
- How?
 - Many methods (vary amplitude, frequency, phase)
 - Focus here for the most part on **A**mplitude **M**odulation



Modulation

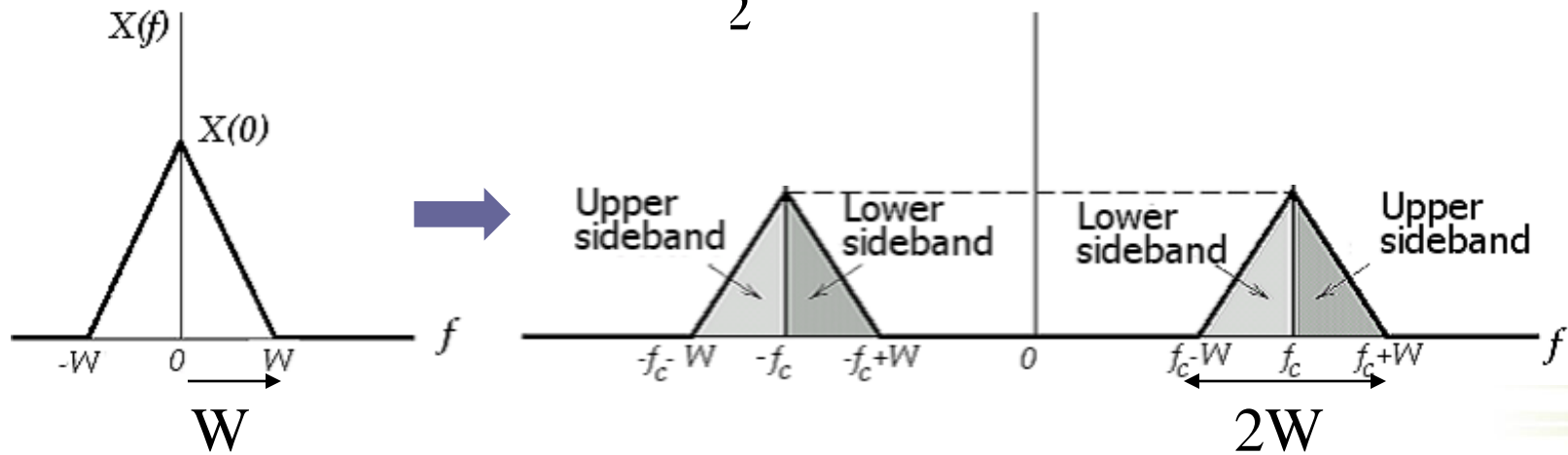
Consider a message signal $x(t)$ has frequency spectrum $X(f)$

In modulation, the message is multiplied (mixed) by a carrier

$\cos(2\pi f_c t)$

$$x(t) \leftrightarrow X(f) \quad (\text{Fourier transform pair})$$

$$x(t) \cos(2\pi f_c t) \leftrightarrow \frac{1}{2} [X(f + f_c) + X(f - f_c)]$$

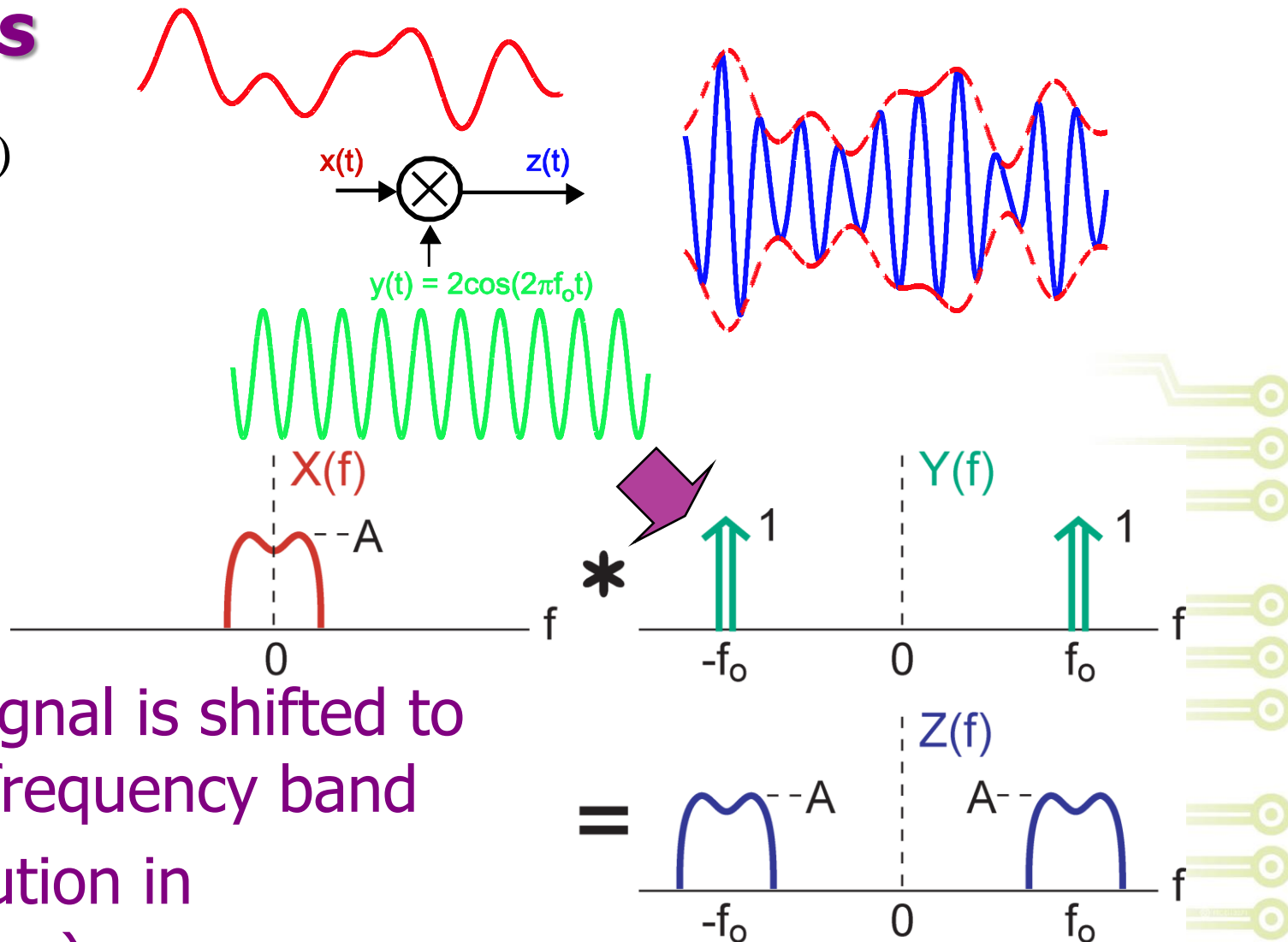


Baseband signal

Frequency translated (modulated) signal

Fourier Transform Allows *Picture* Analysis

(Transmitter)



Understanding modulation

- The idea of modulation is to multiply (i.e. mix) a baseband signal with a carrier signal
- Modulation concept can be easily seen through the trigonometric formula (product to sum sin and cosine) assuming a sinusoidal signal

$$2 \cos(2\pi f_s t) \cos(2\pi f_0 t) = \cos(2\pi(f_0 - f_s)t) + \cos(2\pi(f_0 + f_s)t)$$

↑
signal

↑
carrier

↑
diff.

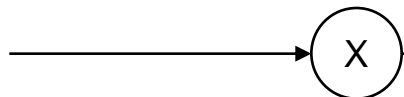
↑
sum

- The signal at f_s is *translated* by a carrier frequency f_0 ($f_0 \gg f_s$). The translation involves both the sum and difference frequency
- Note: higher frequencies lead to smaller antennas

A simple example

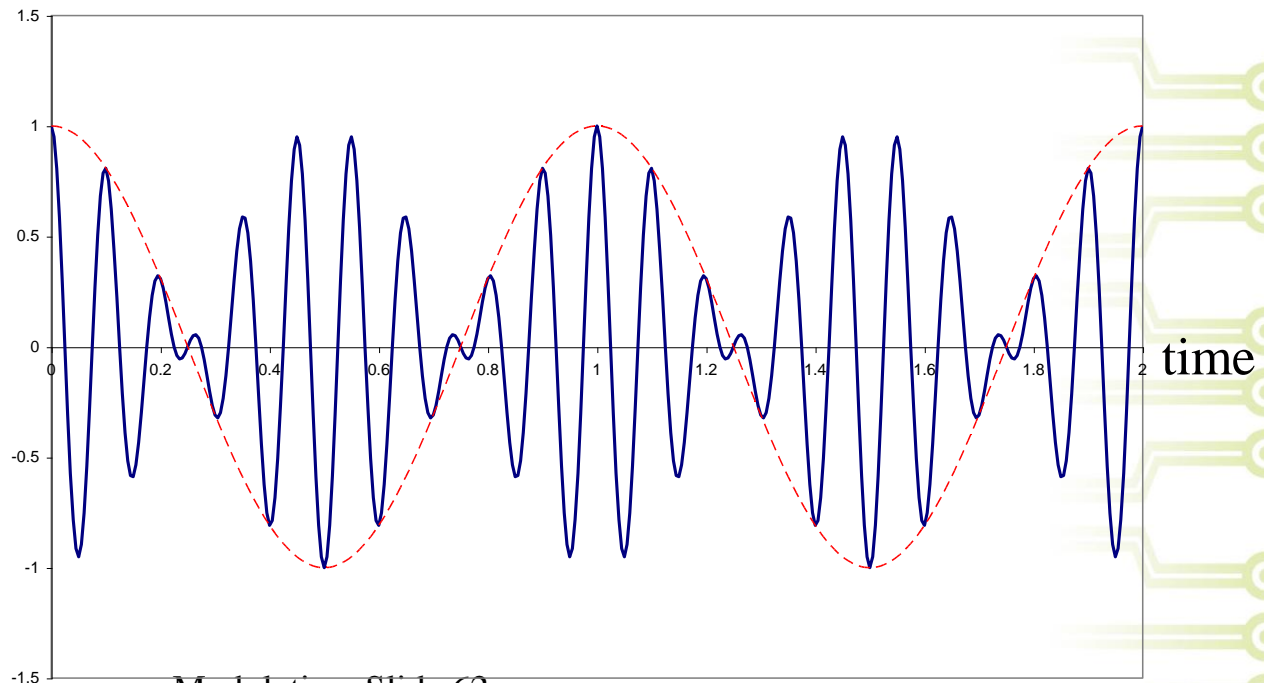
Consider modulating a 1 Hz sinusoidal signal to a carrier frequency at 10 Hz

$$m(t) = \cos(2\pi t)$$



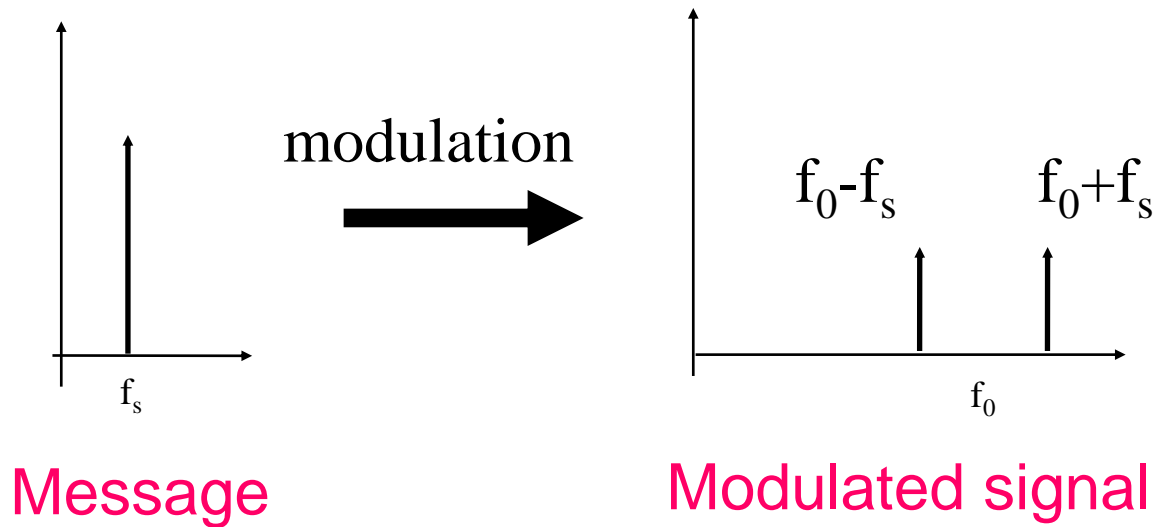
$$\cos(2\pi 10t)$$

$$\cos(2\pi t) \times \cos(2\pi \times 10t) = \frac{1}{2} \cos(2\pi \times 9t) + \frac{1}{2} \cos(2\pi \times 11t)$$



Picture analysis in the frequency domain

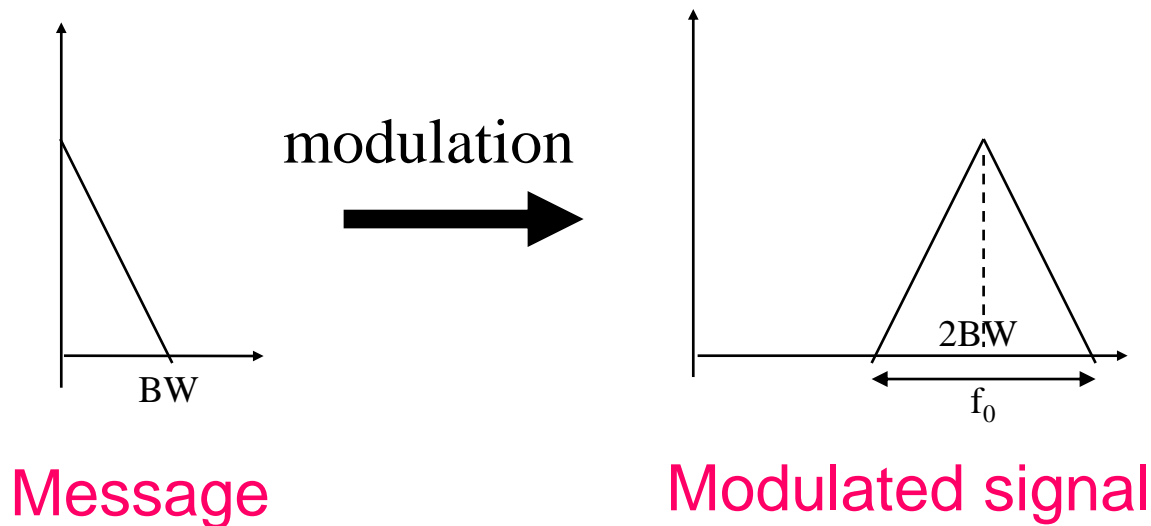
- The message f_s is translated along the frequency axis to $f_0 \pm f_s$



- Note that the bandwidth BW of the message is f_s and that of the modulated signal is $2f_s$

Picture analysis in the frequency domain

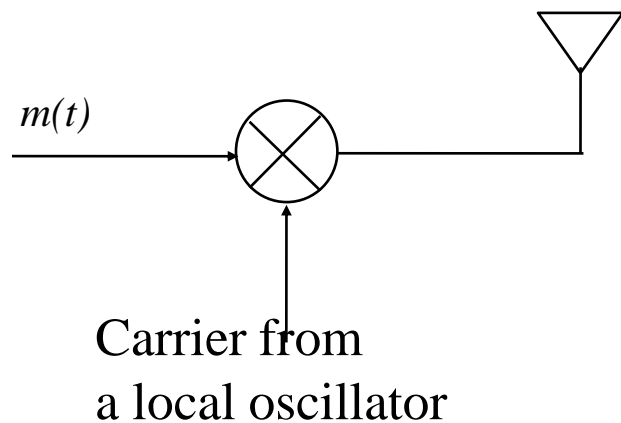
- A message with continuous spectrum and bandwidth BW is translated along the frequency axis to $f_0 \pm BW$



- The modulated signal has a bandwidth *twice* that of the baseband signal

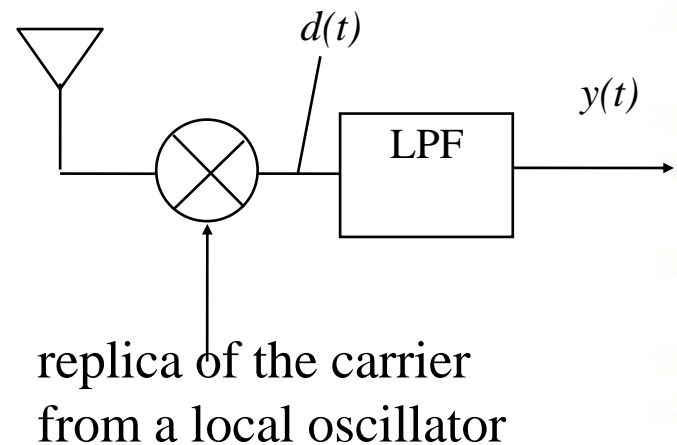
Demodulation

- Demodulation is the process of recovering the original message from the modulated signal
- Demodulation is performed by (i) **mixing** the modulated signal with a replica of the carrier and (ii) **low pass filtering (LPF)**



Modulator

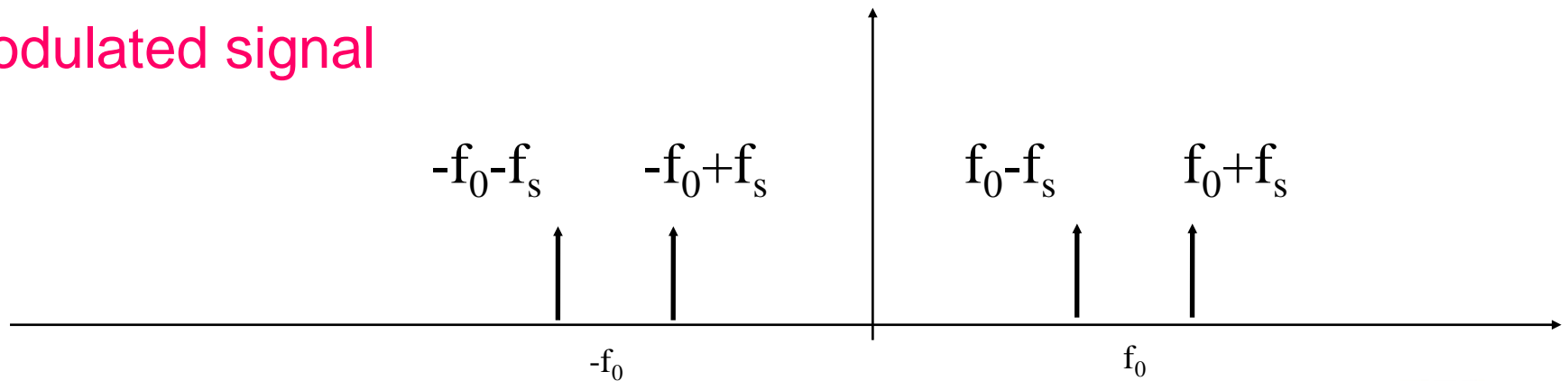
Channel



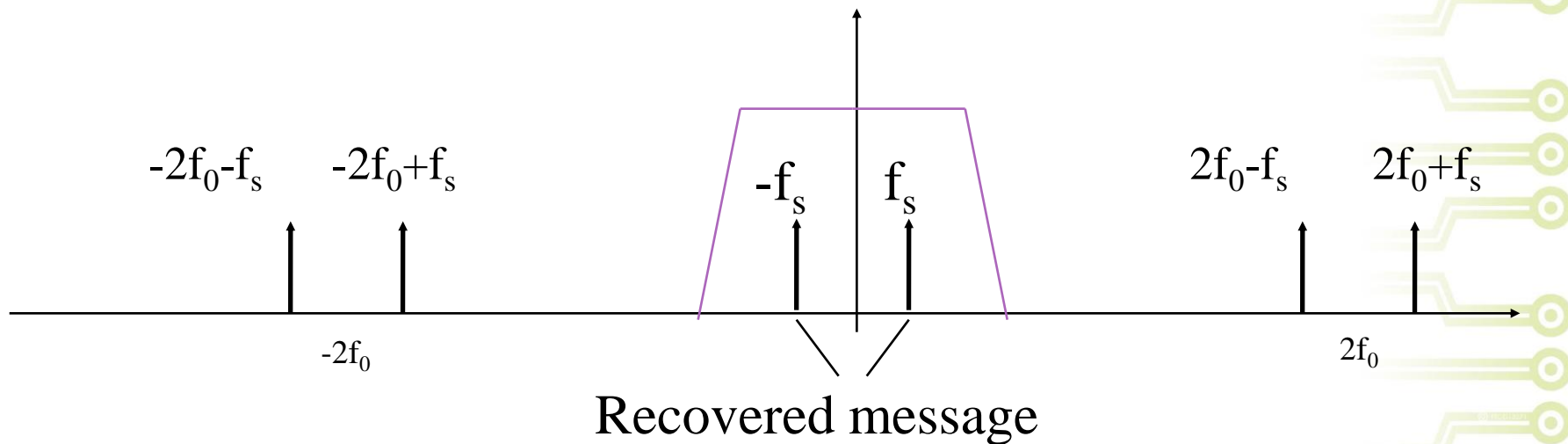
Demodulator

Picture of demodulation

Modulated signal

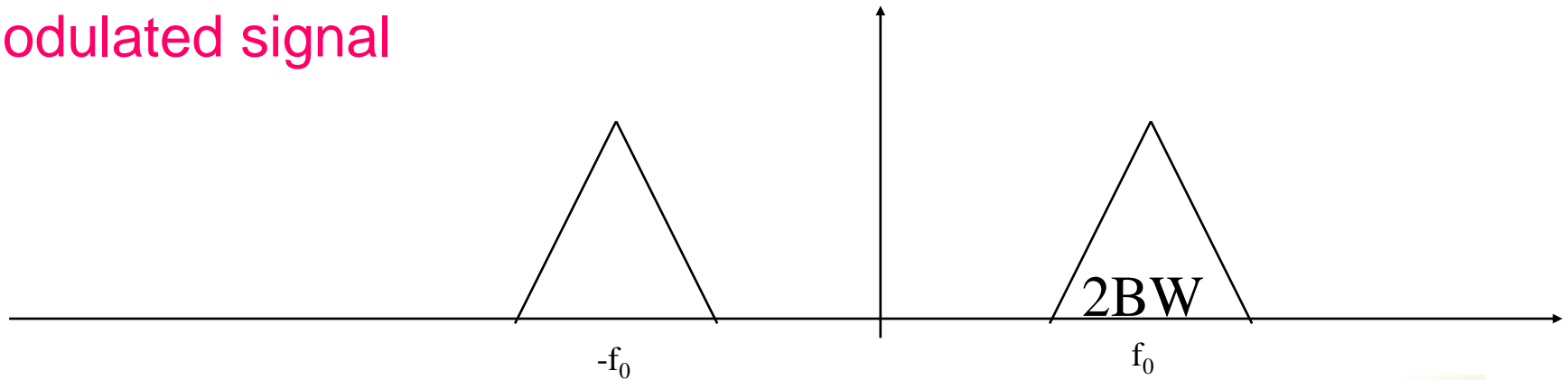


Demodulation (freq. mixing + LPF)

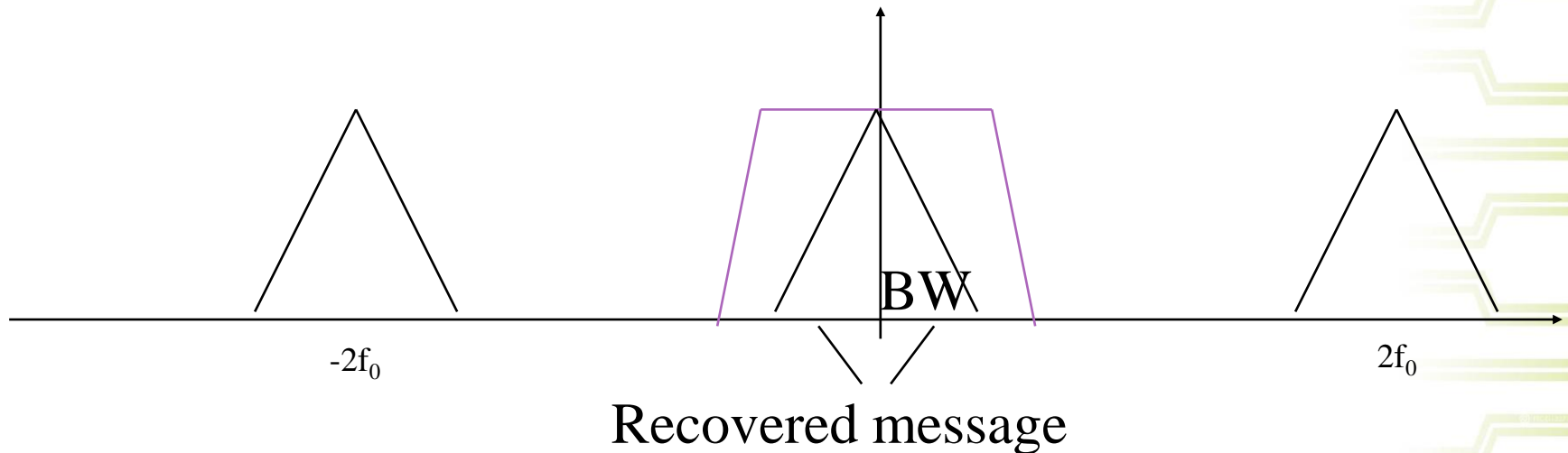


Picture of demodulation

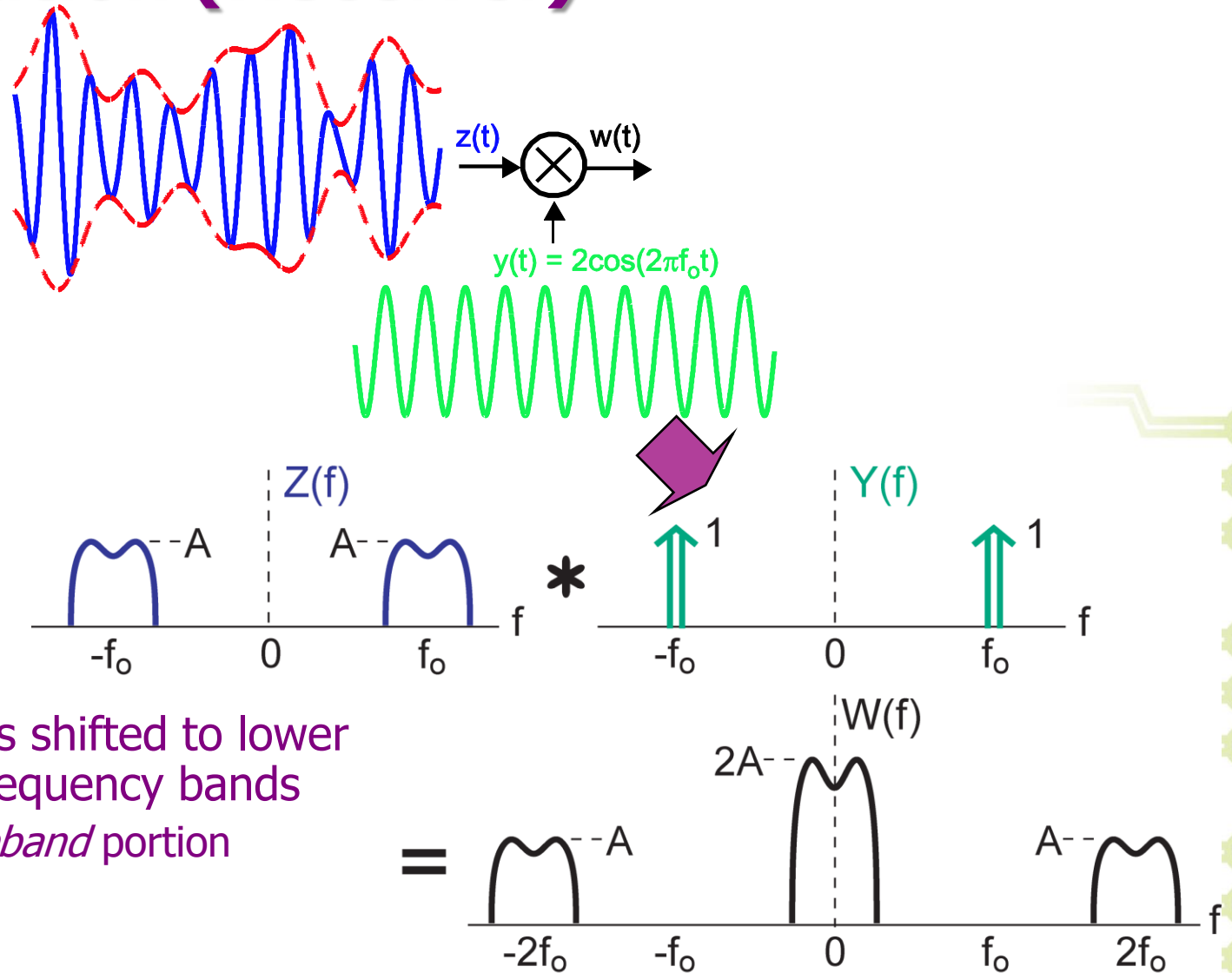
Modulated signal



Demodulation (freq. mixing + LPF)

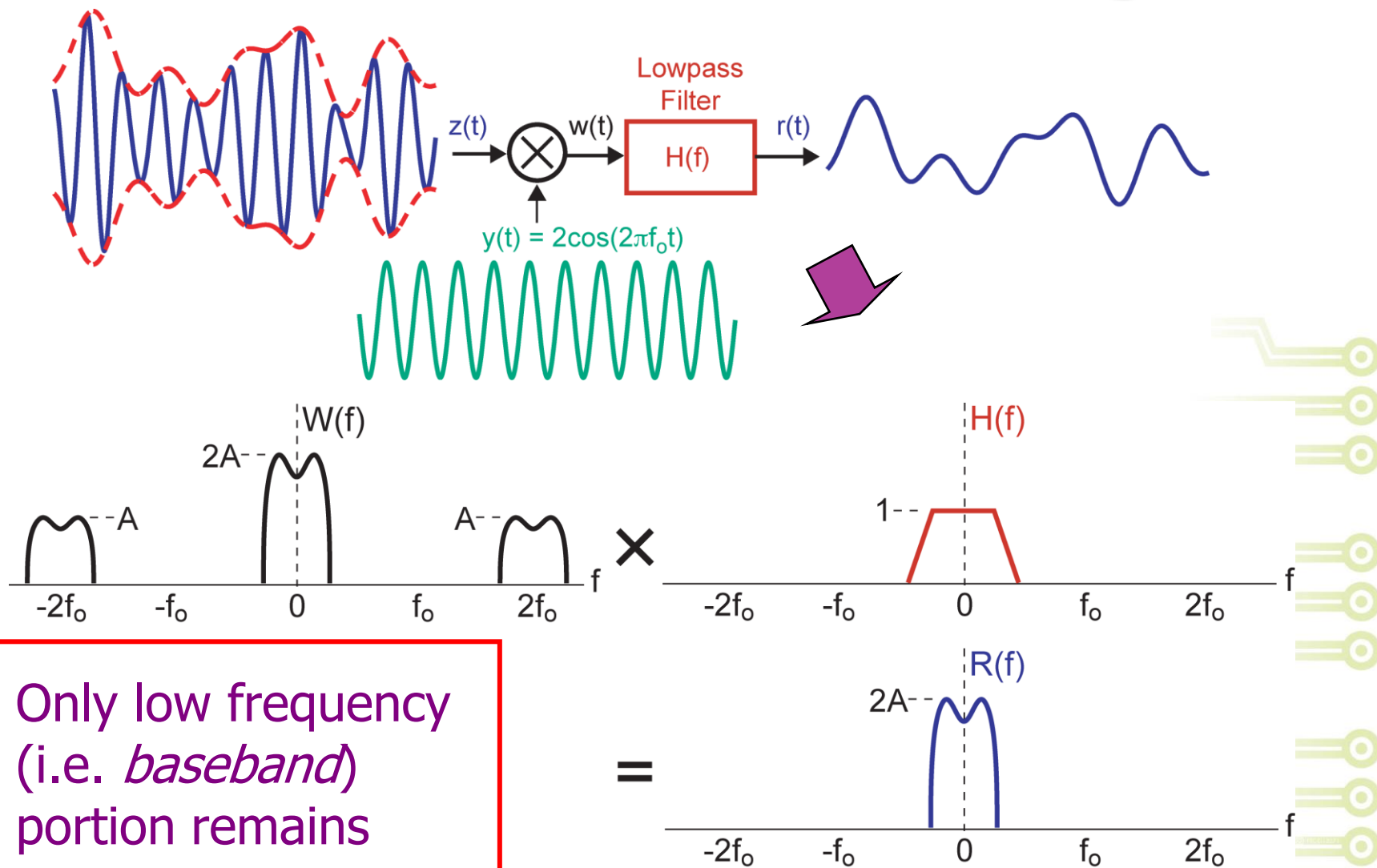


Demodulation (Receiver)



- Input signal is shifted to lower *and* higher frequency bands
 - Want *baseband* portion

Demodulation and Lowpass Filtering



- Only low frequency (i.e. *baseband*) portion remains

Modulation for Acoustics

- Shift the frequency up to 5-10KHz in order to avoid the interference
- Wideband communication since carrier is nearly same as bandwidth
- Perhaps we can use a baseband approach without using a carrier at all

Digital Communications: Relation to Acoustics and Next Steps



Outline- Key Concepts

- Features of Acoustic Water Column Channel
- Digital Communication
 - Communication without channel impairments
 - Channel Noise Impairments
 - Channel Attenuation and Fading Impairments
 - Channel Doppler Impairments
 - Channel Length Impairments
 - Channel Bandlimited Impairments
 - Modulation
- Key highlights
- Next Steps



Features of Acoustic Water Column Channel

- Bandwidth is approx. 0-100KHz
 - Wideband and baseband channel- not narrowband
- Very Long channel
 - For example over a 300m pipe could expect delay spread of 0.5-1 sec
 - Symbol of 0.1ms implies 5000-10000 channel symbol length
- Attenuation
 - Not well characterized but perhaps km propagation range possible

Features of Acoustic Water Column Channel

- Channel variations could be high
 - Carrier sync issues- relative frequency shift is very high compared to wireless?
 - Channel not pseudo-stationary: delay spread larger than coherence time?
 - c is low (1000m/s) so even small speed variations can provide large relative Doppler- $f' = (c+s)/c f$ where Rx approaching
 - For $s=1\text{m/s}$ relative Doppler frequency of 0.1% or 1Hz for 10KHz carrier
 - Moving reflector could be motor or valve
 - Wave speed could also change due to medium speed- water motion
- Noise and Interference
 - Not well characterized- interference at low frequencies such as vehicles and pumps

Next Steps

- Experiments needed for determining Doppler shift- channel variability- probably cannot be modelled without experimental results
- Experiments for Noise and Interference PSD- not just at low frequencies but also in the thermal noise region
- Channel length-
 - Modelling of straight pipes probably ok but need to meld numerical and modal models together
 - Will need a model for in-network pipes- experiments needed!
- Experimental results for attenuation and fading
- Based on these propose a communication systems
- Beware: These issues will have their counterpart in Task 3

Communication Challenge

- If the channel length is larger than the coherence time of channel (channel highly variable case) then it is a very difficult problem
- Approaches
 - Use multiple sensors to separate the modes and therefore effectively reduce channel length- MIMO or space-time equalizer
 - The channel variability may be small in power compared to underlying channel- perhaps can handle as a background phase noise?
- If the channel length is smaller than coherence time then many standard techniques can be used
 - MIMO, OFDM, equalization
- My intuition is that channel length is less than coherence time

Previous Work

- Large body of work for acoustics in open bodies such as oceans and rivers
- Shallow water case more relevant
- Very little performed on acoustics in pipes
 - G. Kokossalakis, Acoustic data communication system for in-pipe wireless sensor networks, PhD Thesis, MIT. [online], 2006
- Push now to look into it due to the need for communication along pipelines for not only water applications but also gas and oil

Previous Results

- Use binary and 4-QAM modulation, Reed Solomon coding on a single carrier with Decision Feedback Equalizers (DFE)
- Carrier frequencies of between 3-63KHz at bandwidths of 2-20KHz were investigated with bit rates of 0.5-21kbps
- Bandwidth efficiency of 0.5-1.05 b/s/Hz were achieved.
- Pipes of 0.15 and 1m radius were considered and equalizers with 20 tapes and distances of up to 500m were simulated.

Previous Results

- Experimental results were also obtained but using air as the propagation medium and PVC pipes of 100mm diameter of up to 10m in length as no water laboratory was available for use.
- To allow comparisons between their experiments and the water results, scaling factors were derived and proposed.
- Straight, bent and branched pipes were also considered.
- They conclude that their proposed system is capable of effective transmission and demonstrated these at equivalent frequencies of 8.7KHz in water.

Relation to Multi Mode Fiber Channels

- Significant similarity to optical MMF Channels
- Propagation as modes and exploiting them to increase bit rates
- Channel modelled as concatenation of sections statistical model can be obtained
- Divide channel into time varying part and slow varying part
Techniques include FDE, OFDM and MIMO
- However acoustics has many differences including noise, interference, attenuation as well as the nature non-linearities

Networking Consideration

- Range of one-hop in-pipe communications is limited → require multi-hop communications to form a network
- Lack of centralized coordinator
 - → requires autonomous, distributed and adhoc networking / multi-access
- Challenges
 - Conventional Wireless Networking Protocols cannot be easily applied
 - WiFi → CSMA/CA
 - Ethernet → CSMA/CD
 - They both fail if the propagation delay is very long!
 - Propagation delay is \ll frame duration in wireless (speed of light)
 - Propagation delay in acoustic pipes can be large (relative to frame duration).



Conclusion

- Acoustic Communication in pipes could open up a new communication technology that is very valuable
- Very little previous work in the area
- Challenges include channel, noise and interference modelling
- May be able to leverage MMF approach
- Could employ advanced communication techniques
- Would welcome your comments and thoughts!



Thanks to many- Interdisciplinary Effort

- Civil Engineering
- Electrical & Electronics Engineering
- Mechanical Engineering
- Mathematics

