Overview of Wireless Communications

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March 2004

Course Nuts and Bolts

Lectures: 22 March – 1 April, daily, 10am – 12pm, RSISE Seminar Room

People:

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Course Assessment

Lectures: 22 March – 1 April, daily, 10am – 12pm, RSISE Seminar Room

Homework: None!

Examination: Class Presentation

- 10 minute presentation on a topic in Wireless Communications
- Talk should be technical in nature and well researched
- Examples: review one of the publications listed, relate a subject in wireless communications to your research....

Course Syllabus

1. Wireless Communications Channels (Mon,Tue) — Tony Pollock
2. Multi-user and Multi-access Channels (Wed,Thur) — Mark Reed
3. Multiple Antenna Channels (Fri, Mon) — Leif Hanlen
4. Mobile Networking (Tue) — Mark Reed
5. Emerging Technology (Wed,Thur) — Dhammika Jayalath

Overview of Wireless Communications

- Wireless Communications Channels -

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March 2004

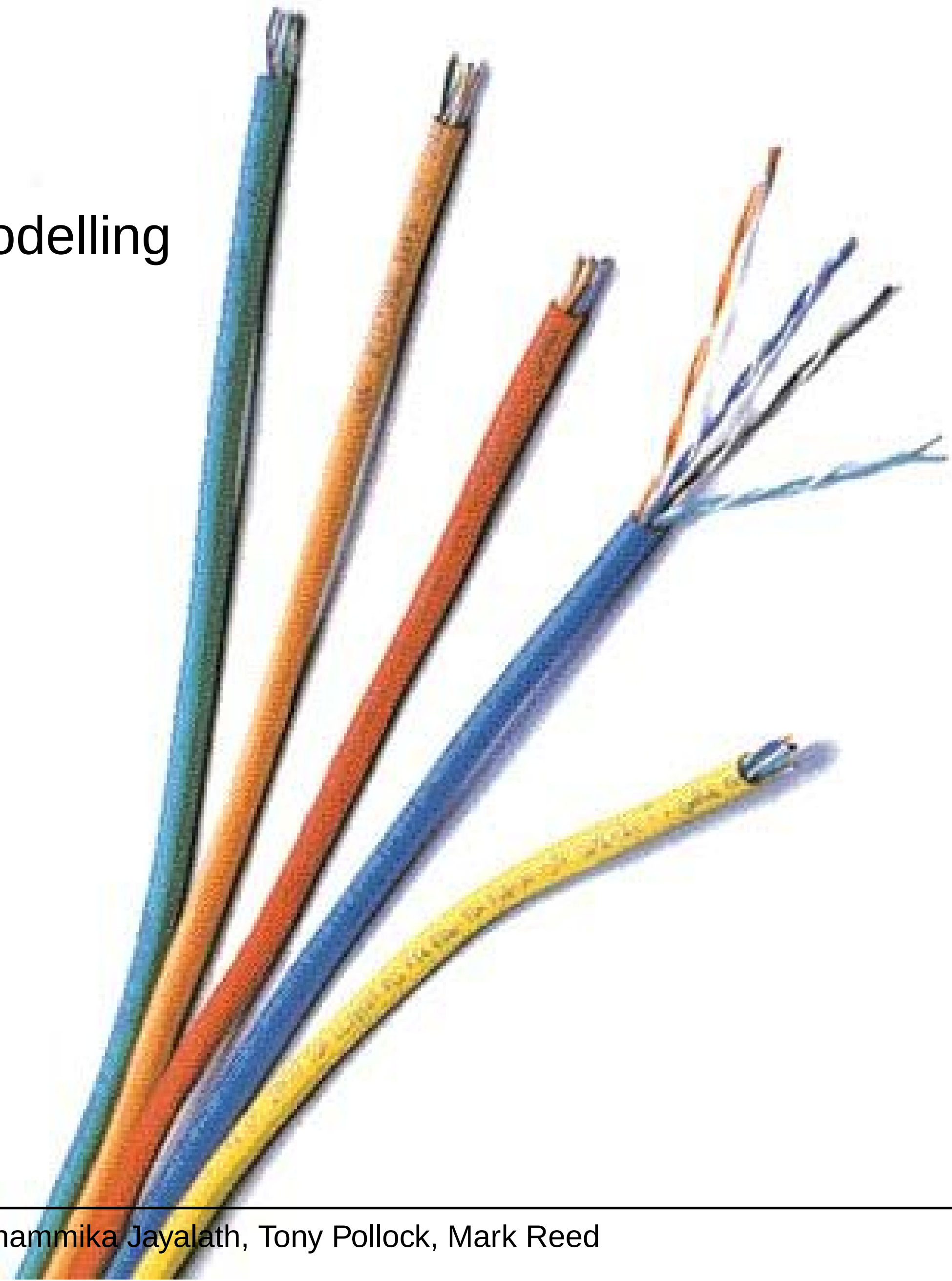
Outline

Wireless Channels :

- Wired vs Wireless
- Channel Characteristics and Modelling
- Channel Capacity

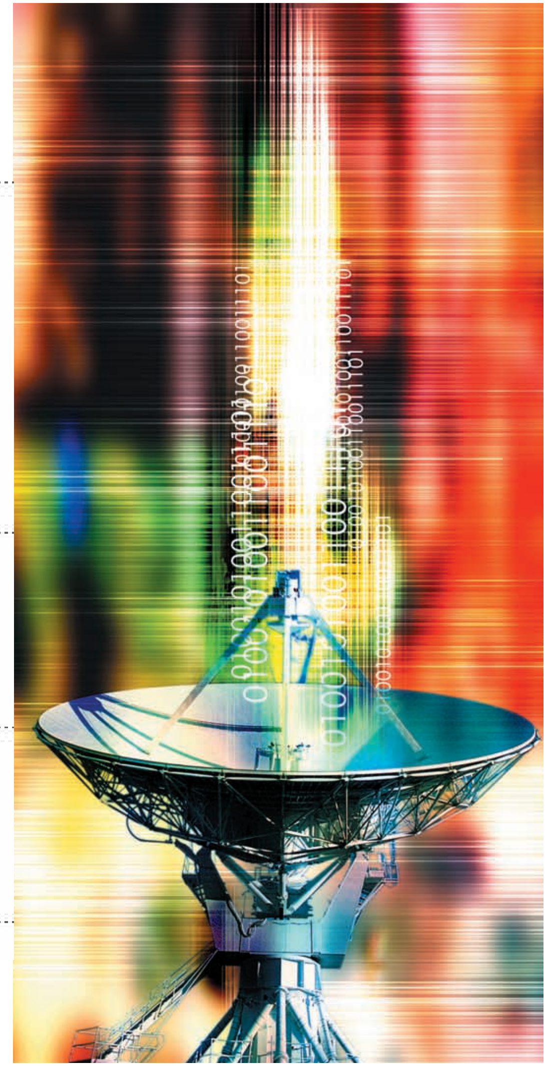
Communications Systems :

- Channel Coding and Decoding
- Modulation and Demodulation
- Equalization



Why Wireless

Ubiquitous Communication Among People and Devices



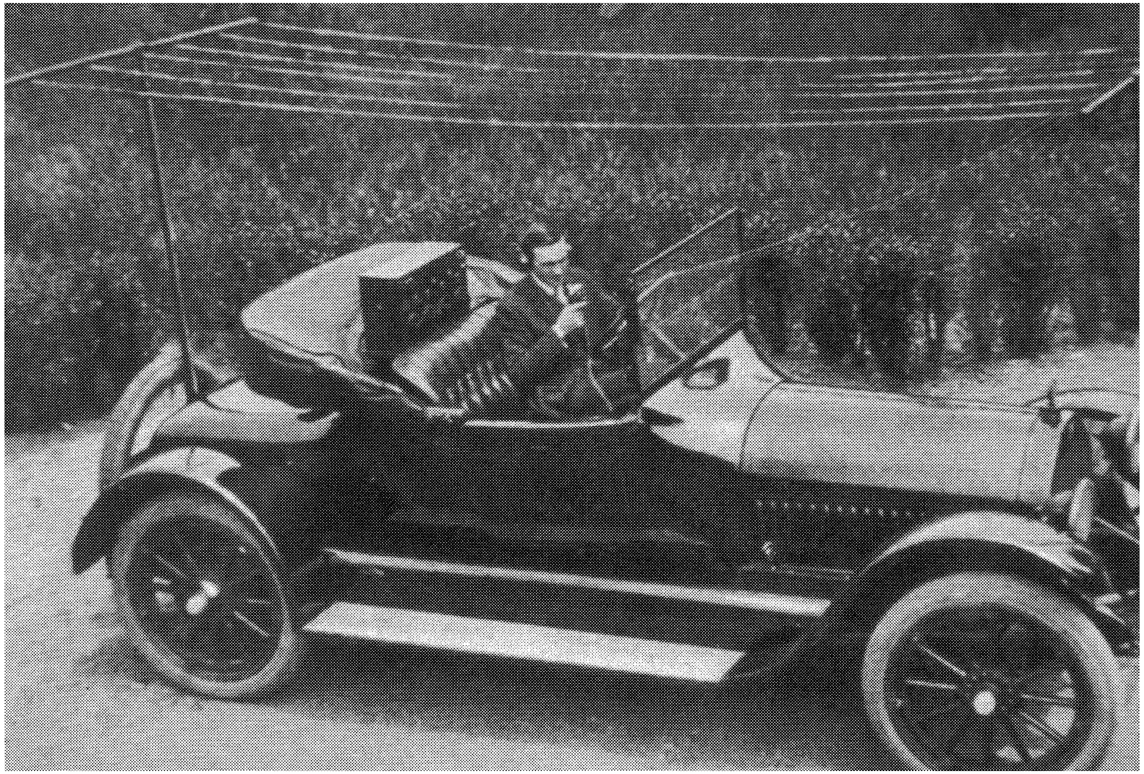
Wireless History

1873	Maxwell predicts the existence of electromagnetic waves
1888	Hertz demonstrates radio waves
1897	Marconi demonstrates mobile wireless communication to ships
1924	US police first use mobile communications
1945	Arthur C. Clarke proposes geostationary communication satellites
1957	Soviet Union launches Sputnik 1 communication satellite
1969	Bell Laboratories in the US invent the cellular concept
1979	NTT cellular system (Japan)
1988	JTACS cellular system (Japan)
1983	AMPS cellular frequencies allocated (US)
1985	TACS (Europe)
1991	USDC (US)
1991	GSM cellular system deployed (Europe)
1993	DECT and DCS (Europe)
1993	PHS cordless system (Japan)
1995	IS95 CDMA (US)
1998	Iridium global satellite system launched
2002	IMT-2000 third-generation cellular mobile systems deployed

1896: Guglielmo Marconi

- Demonstrated concept of wireless telegraphy.
- Built on work of Maxwell and Hertz to send and receive Morse Code.
- Based on long wave ($\gg$ 1 km), spark transmitter technology, requiring very large, high power transmitters.

Wireless vs Mobile



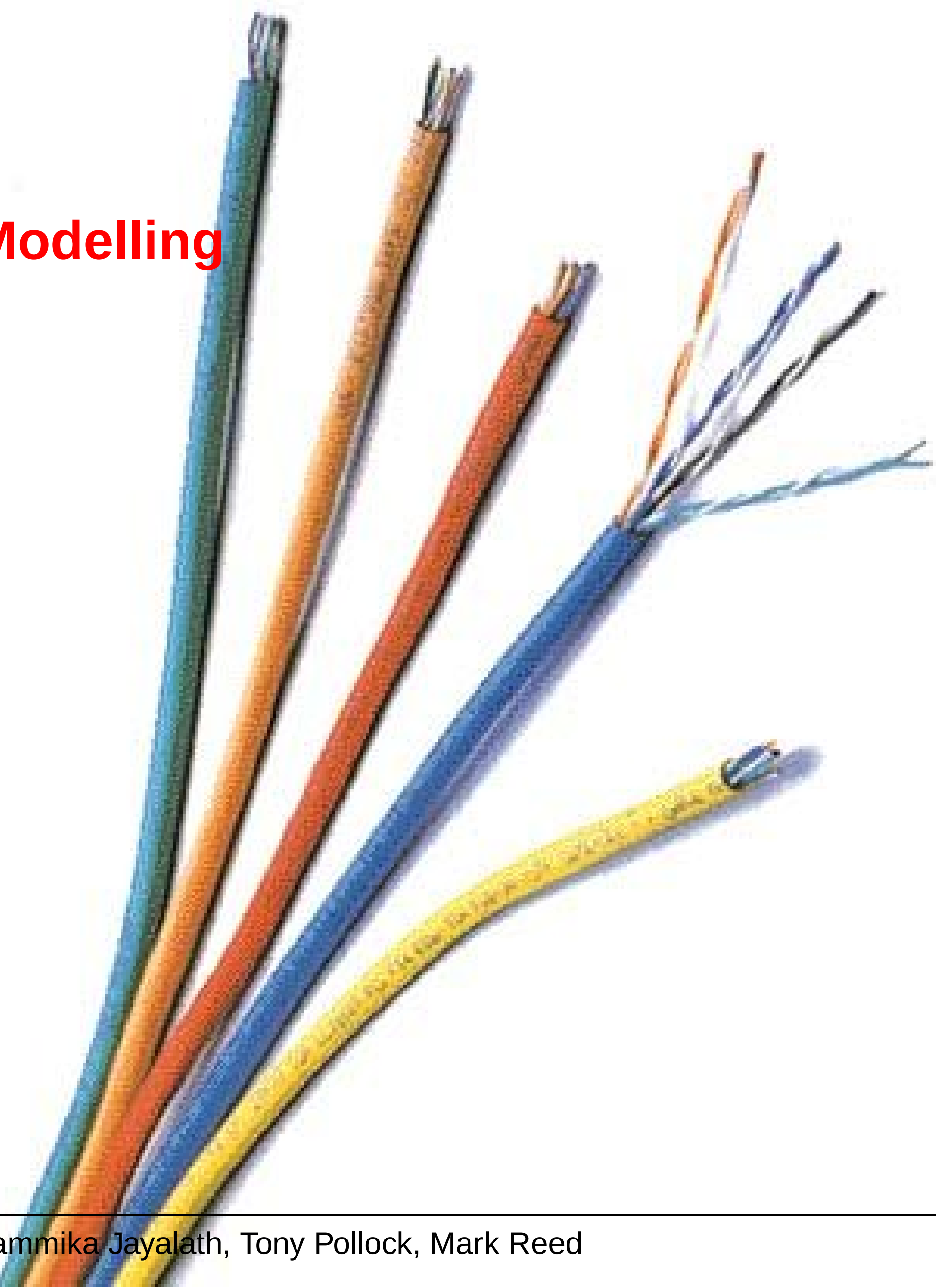
Outline

Wireless Channels :

- Wired vs Wireless
- **Channel Characteristics and Modelling**
- Channel Capacity

Communications Systems :

- Channel Coding and Decoding
- Modulation and Demodulation
- Equalization



Wireless Channels

Channel Characteristics and Modelling :

- Wireless Environment
- Large-Scale Path Loss
  - Free Space Model
  - Reflection
  - Diffraction
  - Scattering
- Small-Scale Fading
  - Flat Fading
  - Frequency Selective Fading
  - Fast Fading
  - Slow Fading
  - Fading Models

Wireless Environment

The wireless channel places fundamental limitations on the performance of wireless communications systems.

Unlike fixed or wired channels which are stationary and predictable, wireless channels exhibit an extremely random nature and are often difficult to characterize and analyze.

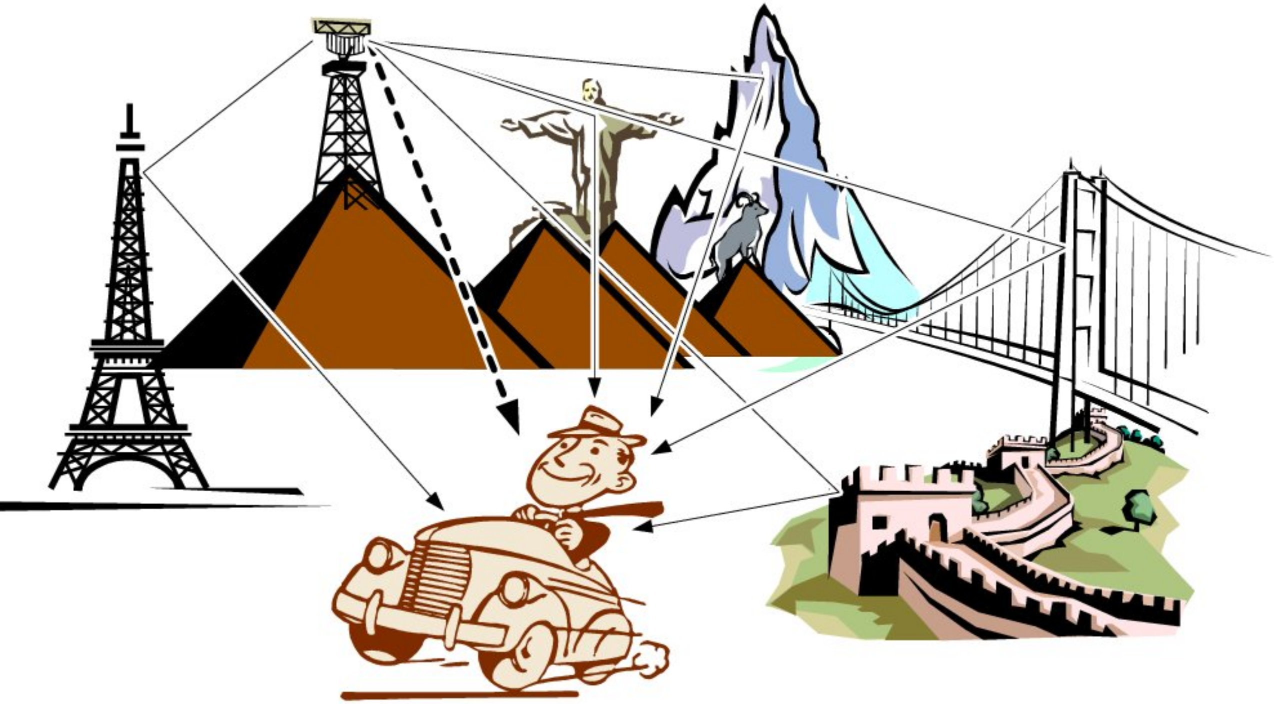
Attenuation: low signal-to-noise ratio (SNR) due to a decrease in electromagnetic energy at receiver (due to distance).

Intersymbol Interference (ISI): distortion of desired symbol due to delay spread (the transmitted symbol is spread over other symbol periods).

Doppler Shift: relative velocities of transmitted and receiver causes frequency shifts in the arriving signal.

Multipath Fading: fluctuations in amplitude, phase, and angle of received signal due to multiple paths the propagating transmitted signal can take to reach the receiver.

Multipath Fading



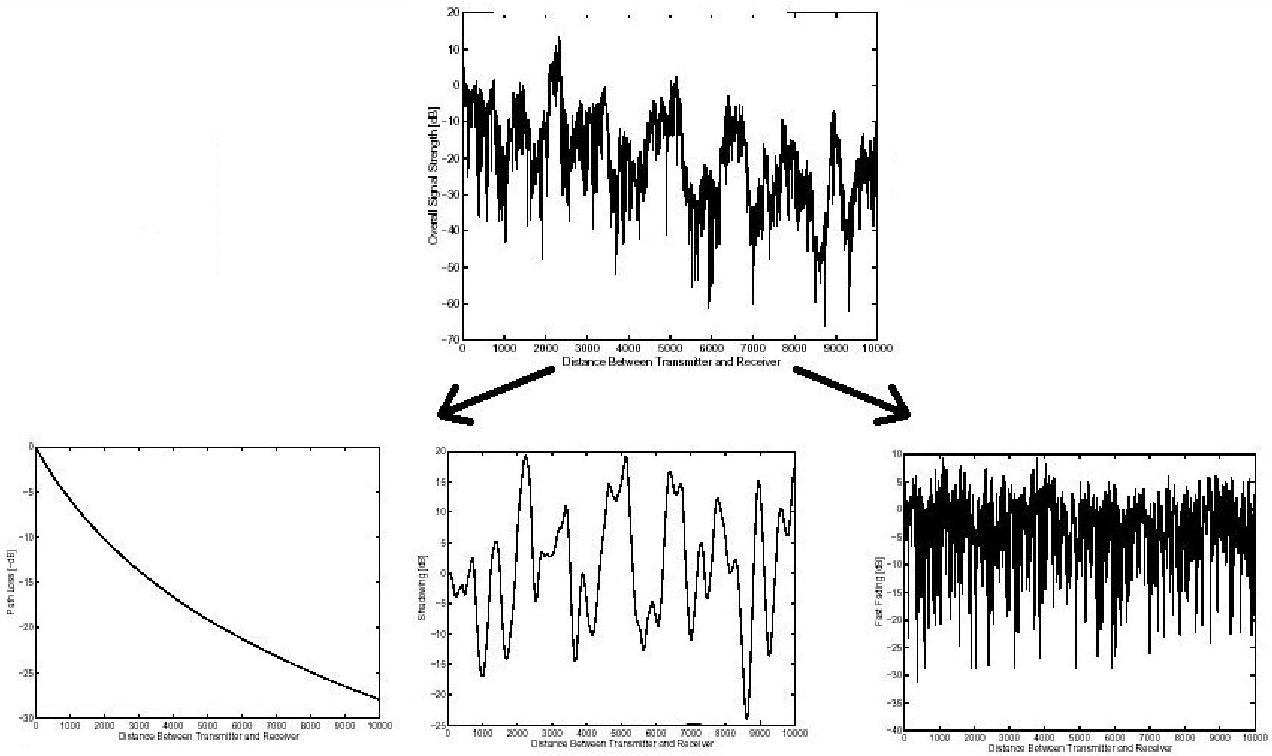
Electromagnetic waves reflecting off or diffracting around objects can result in the signal travelling over multiple paths from the transmitter to the receiver.

Multipath Fading: Multipath propagation can cause fluctuations in the received signals amplitude, phase, and angle of arrival.

Large-Scale and Small-Scale Fading

Large-Scale: characterizing the signal strength over large distances or time.

Small-Scale: characterizing rapid fluctuations of the received signal strength over very short distances or time durations.



Large-Scale

Small-Scale

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Large-Scale Fading

Propagation models which predict the mean signal strength for an arbitrary transmitter-receiver separation distance.

Four basic propagation models:

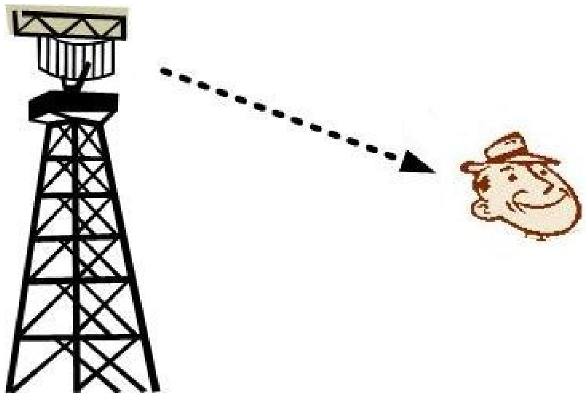
Free Space Propagation: Line-of-sight path between transmitter and receiver.

Reflection: when the propagating electromagnetic wave impinges upon an object which has large dimensions compared to the wavelength of the propagating wave.

Diffraction: when the path between transmitter and receiver is obstructed by a surface with sharp irregularities such as edges.

Scattering: when the propagating wave impinges upon objects with dimensions that are small compared to its wavelength, and the number of objects per unit volume is large.

Large-Scale Fading: Free Space Propagation



Received power decays as a function of separation d.

Friis equation: (farfield) free space received power P_{R}

$$P_{\mathsf{R}}(d) = \frac{P_{\mathsf{T}} G_{\mathsf{T}} G_{\mathsf{R}}}{(4\pi d/\lambda)^2 L}$$

P_{T}, G_{T} are transmit power and antenna gain, G_{R} is receiver antenna gain, and L is a system loss factor ($L \geq 1$).

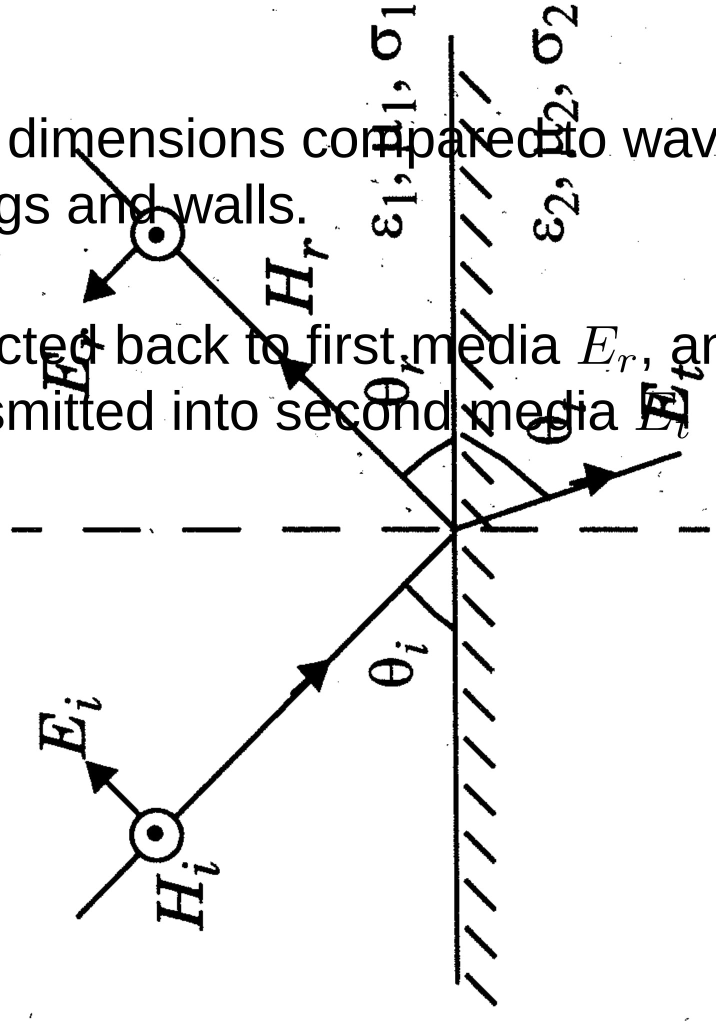
Path Loss: signal attenuation

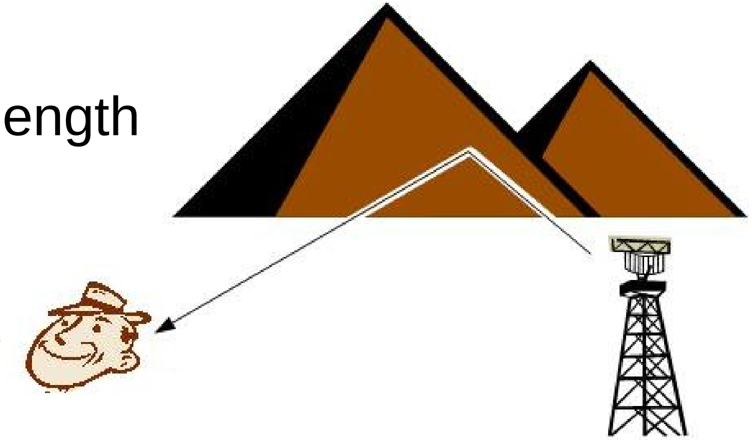
$$\text{Path Loss (dB)} = 10 \log_{10} \frac{P_{\mathsf{T}}}{P_{\mathsf{R}}}.$$

Large-Scale Fading: Reflection

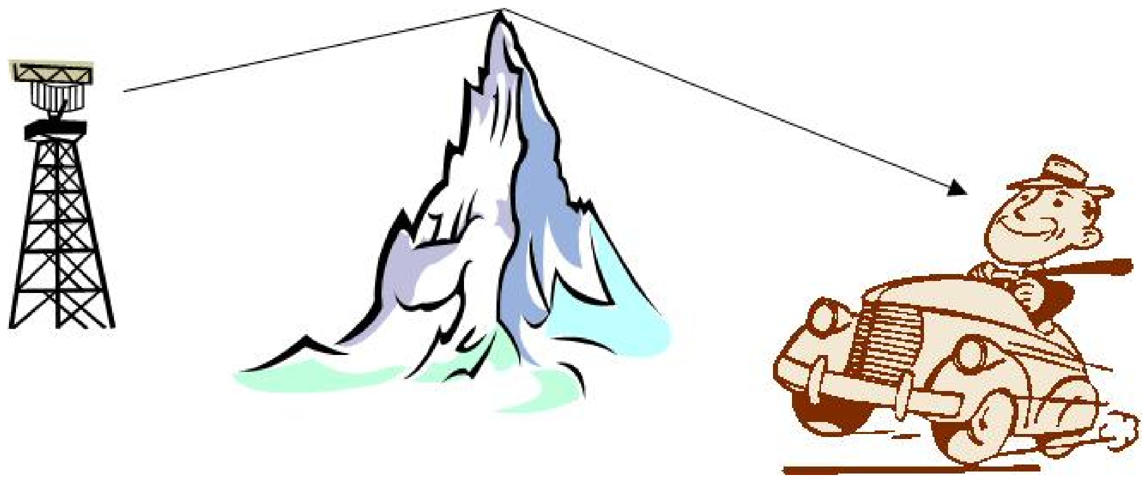
Object has very large dimensions compared to wavelength such as earth, buildings and walls.

Part of energy is reflected back to first media E_r, and part of energy is transmitted into second media E_t





Large-Scale Fading: Diffraction

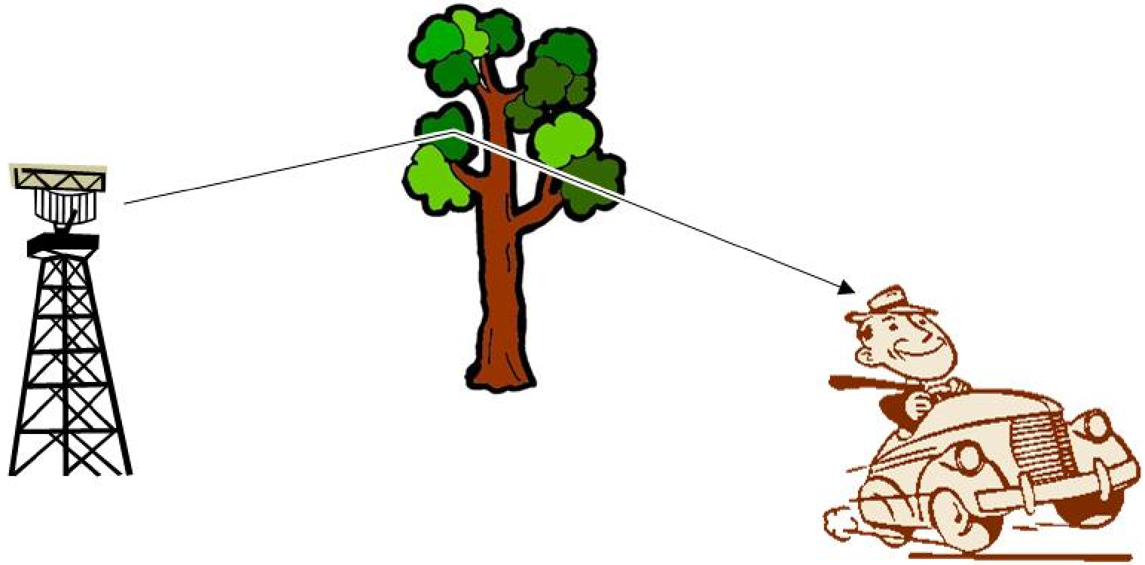


Diffraction allows signals to propagate behind obstructions.

The received field strength decreases rapidly as receiver moves deeper into obstructed (shadowed) region.

Described by Huygens principle (all points on a wavefront can be consider point sources).

Large-Scale Fading: Scattering



Scattered waves are produced by rough surfaces, small objects or other irregularities in the channel.

The roughness of some surfaces often induces propagation effects different from specular reflection.

Scattering gives rise to stronger received signals than predicted by reflection and diffraction models alone, as the reflected energy is spread out (diffused) in all directions.

Small-Scale Fading

Many physical factors in the channel influence small-scale fading:

Multipath: Reflecting and scattering objects in the channel result in multiple versions of the transmitted signal arriving at the receiver, displaced with respect to one another in time and spatial orientation. The random phase and amplitudes of the different multipath cause fluctuations in signal strength inducing small-scale fading and distortion.

Speed of receiver: Relative motion between transmitter and receiver results in random frequency modulation due to different Doppler shifts on each multipath.

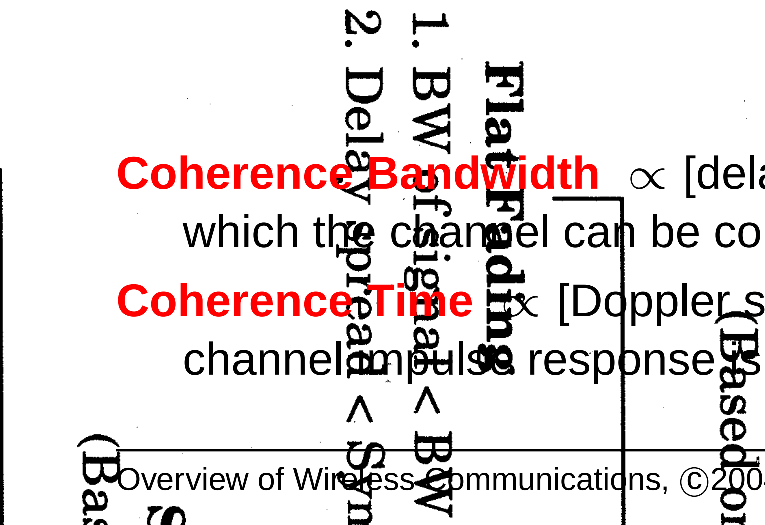
Speed of surrounding objects: If the surrounding objects are in motion and move at a greater rate than the receiver they induce a time varying Doppler shift on multipath components.

Transmitted signal bandwidth: If the transmitted signal has bandwidth smaller than that of the multipath channel then the amplitude of the received signal will change rapidly.

Characterizing Small-Scale Fading

Coherence Bandwidth $\propto$ [delay spread]$^{-1}$: statistical measure of the range of frequencies over which the channel can be considered "flat" (equal gain and linear phase).

Coherence Time $\propto$ [Doppler spread]$^{-1}$: statistical measure of the time duration over which the channel impulse response is essentially invariant.

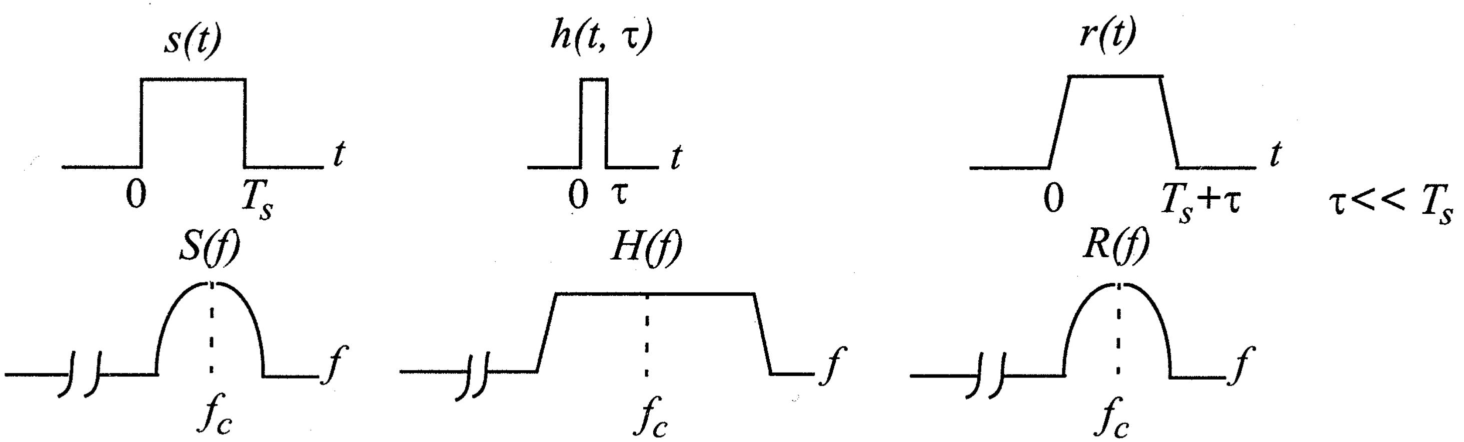


Small-Scale Fading: Flat Fading

- **Flat fading** of the received signal occurs when the channel has constant gain and linear phase response over a bandwidth that is greater than the bandwidth of the transmitted signal.

- For flat fading the multipath channel is such that the spectral characteristics of the transmitted signal are preserved at the receiver.

- The strength of the received signal may change with time, due to fluctuations in the gain of the channel caused by multipath.

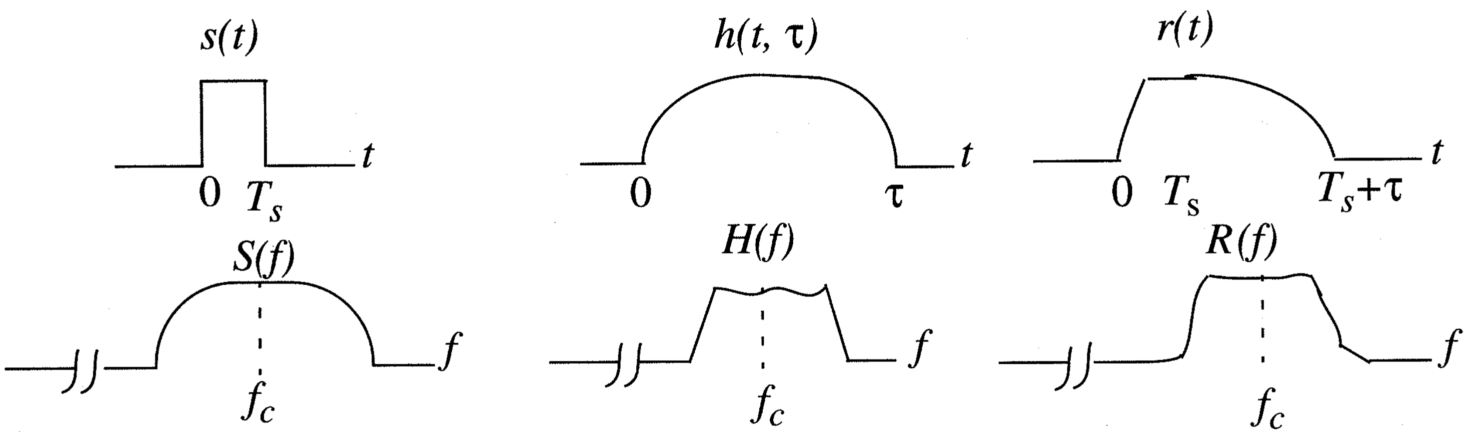


Small-Scale Fading: Frequency Selective Fading

- **Frequency selective fading** of the received signal occurs when the channel has constant gain and linear phase response over a bandwidth that is smaller than the bandwidth of the transmitted signal.

- The received signal includes multiple versions of the transmitted waveform which are attenuated (faded) and delayed in time.

- This fading is due to time dispersion of the transmitted symbols within the channel (ISI).



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Small-Scale Fading: Fast Fading

- A **fast fading channel** has an impulse response which changes rapidly within the symbol duration.
- Here the coherence time of the channel is smaller than the symbol period of the transmitted signal.
- This causes frequency dispersion (*time selective fading*) due to Doppler spreading, which leads to signal distortion.

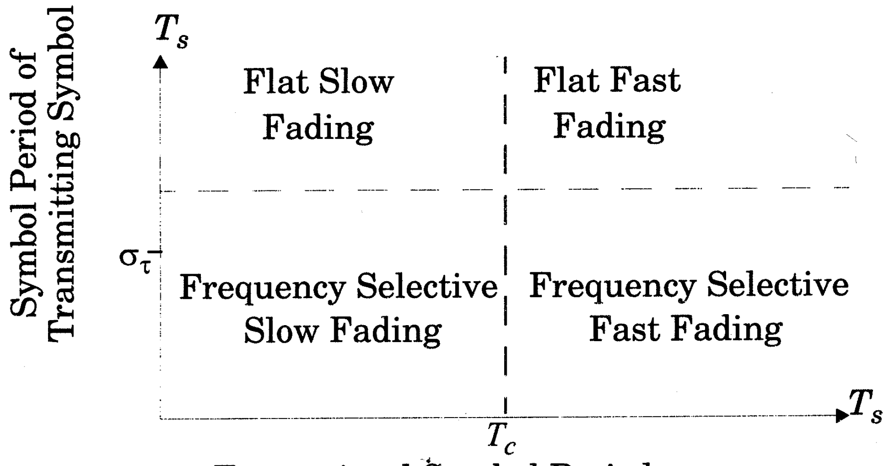
Small-Scale Fading: Slow Fading

- In a **slow fading channel** the channel impulse response changes at a rate much slower than the transmitted baseband signal.

- Whether a signal undergoes *fast* or *slow* fading depends on the velocity of the mobile (or the velocity of objects in the channel) and the baseband signalling.

- Fast and Slow fading deal with the relationship between the time rate of change in the channel and the transmitted signal.

Small-Scale Fading



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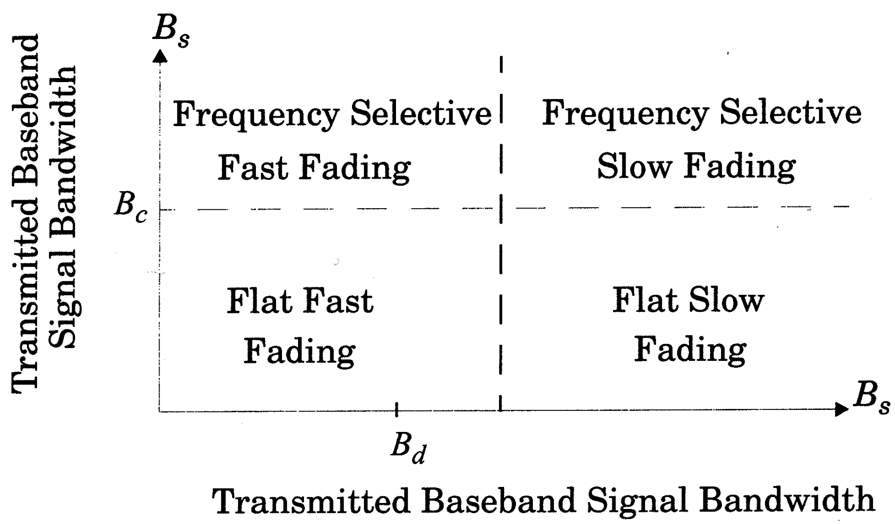

(b)

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Fading-Channel Models

Rayleigh: used to model environments with a large number of reflected and scattered waves.

Nakagami-q: used on satellite links where strong ionospheric scintillation occurs.

Nakagami-n (Ricean): used for similar environments as Rayleigh, except where there is a dominant stationary (nonfading) signal component (such as line-of-sight).

Nakagami-m: used to model fading-channel conditions that are more severe than the Rayleigh distribution. Good model for land-mobile and indoor-mobile multipath propagation.

Fading-Channel Models

Mobile systems with no LOS path between transmitter and receiver antenna, propagation of reflected and refracted paths through troposphere and ionosphere, ship-to-ship radio links.	Rayleigh
Satellite links subject to strong ionospheric scintillation.	Nakagami-q (Hoyt)
Propagation paths consisting of one strong direct LOS component and many random weaker components microcellular urban and suburban land mobile, picocellular indoor and factory environments	Nakagami-n (Rice)
Land mobile, indoor mobile multipath propagation as well as ionospheric radio links.	Nakagami-m
Terrain, buildings, trees urban land mobile systems, land mobile satellite systems.	Log-normal shadowing
Nakagami-m multipath fading superimposed on log-normal shadowing. Congested downtown areas with slow-moving pedestrians and vehicles. Also in land mobile systems subject to vegetative and/or urban shadowing.	Composite gamma/log-normal
Convex combination of unshadowed multipath and a composite multipath/ shadowed fading. Land mobile satellite systems.	Combined (time-shared) shadowed/unshadowed

Combating Channel Impairments

Increase transmitter power: counters flat fading, but costly

(Adaptive) Equalization: compensates for intersymbol interference due to time dispersion

Antenna or space diversity: usually, two (or more) receiving antennas

Forward error correction: transmit redundant data bits - coding gain provides fading margin

Automatic Repeat Request (ARQ): retransmission protocol for blocks of data (e.g. packets) in error

Other: packet length adaptation, spreading gain adaptation etc.

First we need a measure of achievable performance......**Information Theory**

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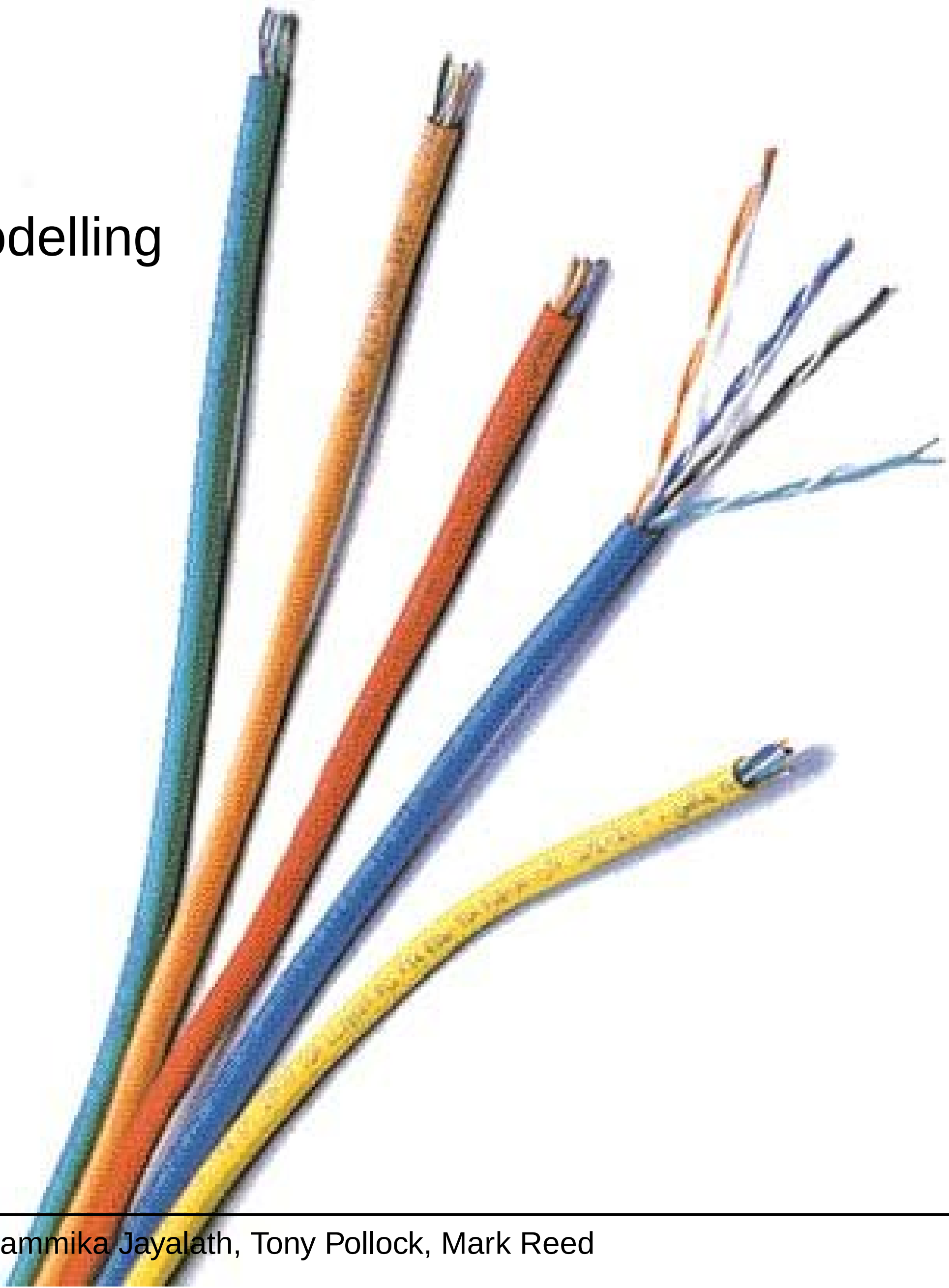
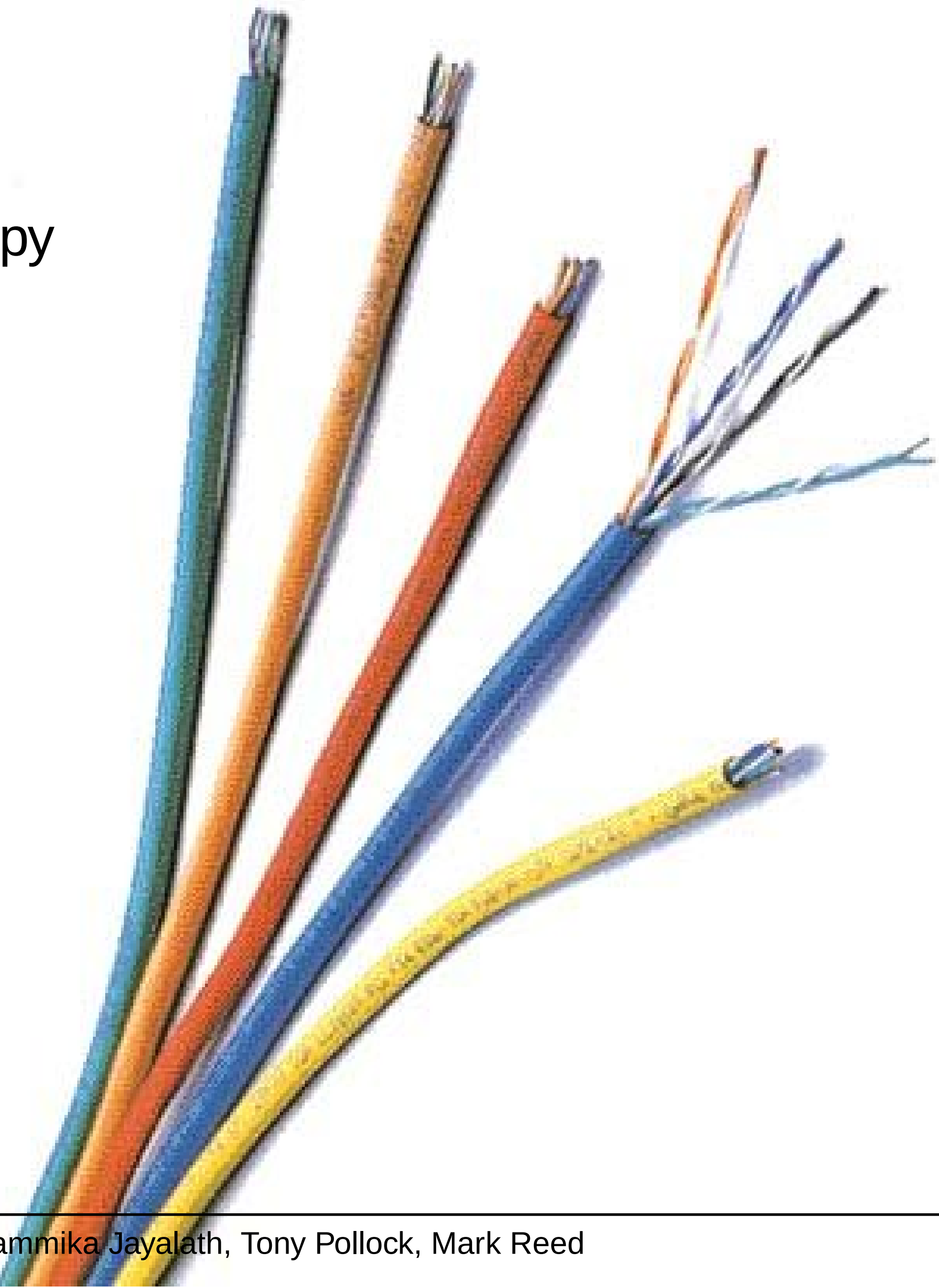


Wireless Channels

Channel Capacity :

- Basics of Communication
- Information? Properties of Entropy
- Channel Coding



Basics of Communications

The basic goal is **efficient** and **reliable** communication in an uncooperative (and possibly hostile) environment.

Efficient: the transfer of information must not require a prohibitive amount of time and effort.

Reliable: the received data stream must resemble the transmitted stream to within narrow tolerances.

These two requirements will always be at odds, and the fundamental problem of communications is to reconcile them as best we can.

Information passes from a source to a sink via a channel. Here we are allowed to choose exactly the way the data stream is structured at the source and the way it is handled at the sink, however, the behaviour of the channel is not in general under our control.

Source Coding

With **source coding** we are concerned with **efficient** communication in an environment we assume is not hostile.

Source coding takes advantage of the *statistical properties* of the original data stream. In data **compaction** and **compression**[1] redundancy is removed in the interest of efficient use of the available message space.

Compaction: coding scheme that allows perfect reconstruction of the original data. e.g. Morse Code - the letter 'e' is the most frequently used in the English language, and is assigned the shortest Morse code message (a single dot).

Compression: coding scheme that allows a close approximation to the original data. e.g. Graphic images - the eye can not see things perfectly, a good likeness of the original is acceptable.

[1] the two terms compaction and compression are used in order to distinguish lossless and lossy compression

Channel Coding

With **channel coding** we are concerned with **reliable** communication in a possibly hostile environment.

To guarantee that the original data can be recovered from a version that is not too badly corrupted, we add redundancy to our message at the source.

One of the oldest forms of coding for error control is the adding of a parity check bit to an information string. Sufficient redundancy allows us not only to detect errors have occurred, but to also fix them

Language is a good example of a sufficiently redundant source so that we can usually recover from imperfect reception.

IF U CN RD THS U CN GT A JB

Information Theory

Fundamental Problem: find a scheme that has both reasonable information content and reasonable error handling ability.

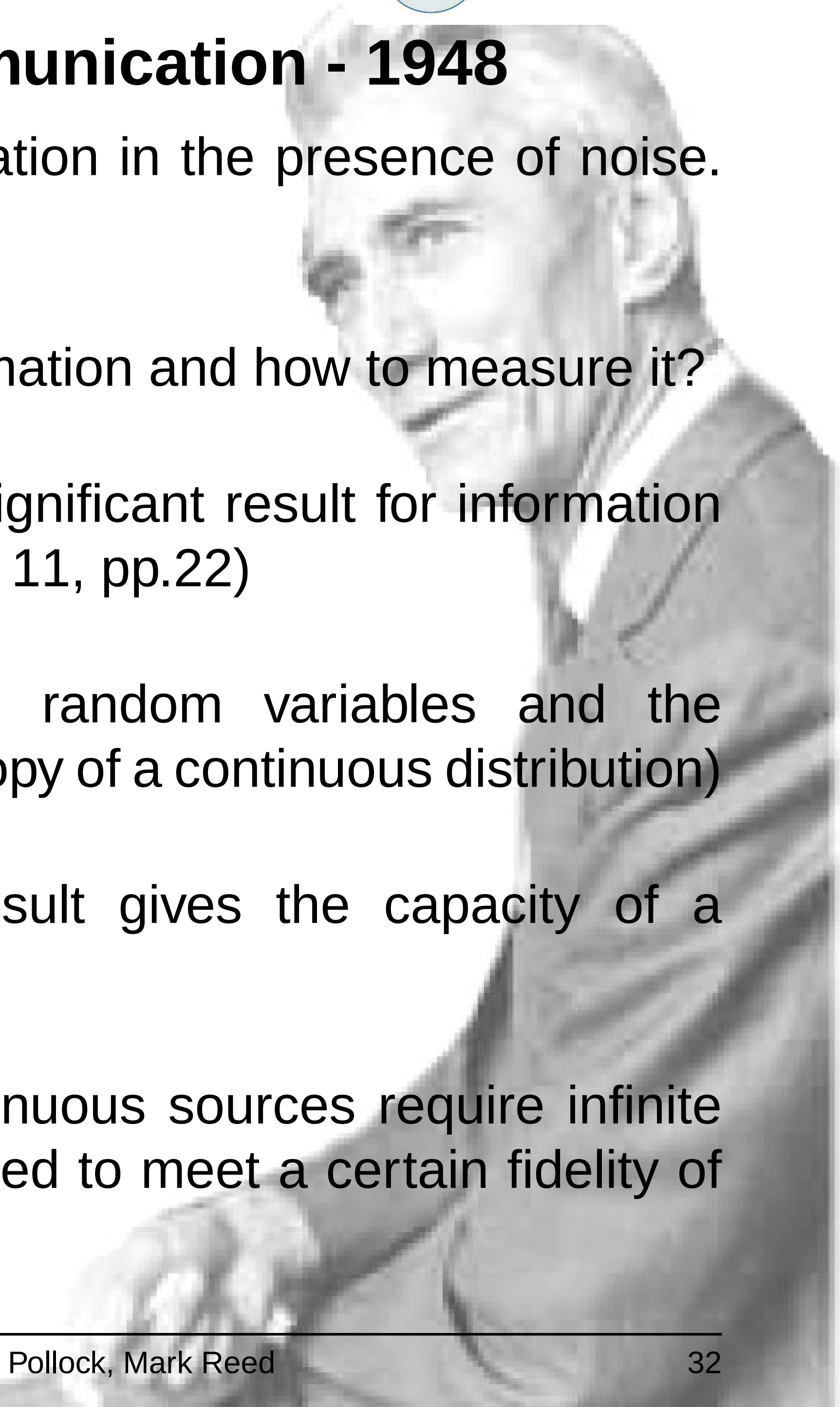
Is this even possible? Well Yes! - shown by the granddaddy of **Information Theory** Claude Elwood Shannon in his 1948 paper.

Shannon's Information Theory enables us to answer the two fundamental questions in communications:

1. What is the ultimate data compression? (efficiency)

2. What is the ultimate transmission rate of communication? (reliability)

A Mathematical Theory of Communication - 1948



Shannon wanted efficient and reliable communication in the presence of noise. His famous article is made up of five parts:

1. **"Discrete Noiseless Systems"** - what is information and how to measure it?
2. **"The Discrete Channel with noise"** - most significant result for information theory, 'Shannon's Second Theorem' (Theorem 11, pp.22)
3. **"Mathematical Preliminaries"** - continuous random variables and the connection with discrete random variables (entropy of a continuous distribution)
4. **"The Continuous Channel"** - significant result gives the capacity of a Gaussian channel
5. **"The Rate for a Continuous Source"** - continuous sources require infinite capacity for exact recovery. What rate is required to meet a certain fidelity of recovery?

Information?

The use of probabilities to describe a situation implies some *uncertainty*.

Suppose we have a device that can produce 3 symbols, A,B, or C. As we wait for the next symbol, we are *uncertain* as to which symbol it will produce. Once the symbol appears and we observe it, our uncertainty decreases, we would consider that we have received some *information*.

We would like to develop a usable mathematical measure of the *information* we get from observing the occurrence of an event having probability p.

We ignore any particular features of the event and only observe whether or not it happened. This allows us to think of the event as the observation of a symbol whose probability of occurring is p.

Therefore we are defining the *information* in terms of the probability p.

We wish our information measure $I(p)$ to have the following properties:

1. Information is a non-negative quantity

$$I(p) \geq 0$$

2. If an event has probability 1, we get no information from observing the event

$$I(1) = 0$$

3. If two independent events occur then the information we get from observing the events is the sum of the two information measures

$$I(p_{12}) = I(p_1 * p_2) = I(p_1) + I(p_2)$$

4. We want $I(p)$ to be a continuous (and monotonic) function of the probability. Small changes in probability should result in small changes in information.

Summarizing: we wish our information measure to have the following:

1. $I(p) \geq 0$
2. $I(1) = 0$
3. $I(p)$ is monotonic and continuous in p
4. $I(p_1 * p_2 * \cdots * p_N) = I(p_1) + I(p_2) + \cdots + I(p_N)$
5. $I(p^a) = a * I(p)$, $a > 0$

we can then derive our measure of information as

$$I(p) = -\log_b(p) = \log_b(1/p)$$

for some positive constant b. Where the base b determines the units we are using.

We can change the units by changing the base, for $b_1, b_2, x > 0$

$$\log_{b_2}(x) = \log_{b_2}((b_1)^{\log_{b_1}(x)}) = \log_{b_2}(b_1)\log_{b_1}(x)$$

that is each base corresponds to different information measures which are just constant multiples of each other.

Common units are:

1. $\log_2$ units are **bits** (‘binary’)
2. $\log_3$ units are **trits** (‘trinary’)
3. $\log_e$ units are **nats** (‘natural logarithm’)
4. $\log_{10}$ units are **Hartleys** (from an early researcher)

We will not bother to specify the base. Unless stated we consider $\log(x) = \log_2(x)$.

Entropy

Suppose now we have n symbols $\{x_1, x_2, \ldots, x_n\}$, and some source is providing us with a stream of these symbols. Suppose further that the source emits the symbols with probabilities $\{p_1, p_2, \ldots, p_n\}$, respectively. For now, we also assume that the symbols are emitted independently (successive symbols do not depend on past symbols).

What is the average amount of *information* we get from each symbol we observe in the stream?

What we really want is a weighted average. If we observe symbol x_i, we will get $-\log p_i$ bits of *information* from that particular observation. In a long run, say N observations, we will see approximately Np_i occurrences of symbol x_i.

For the N independent observations we will get total information I of

$$I = -\sum_{i=1}^{n} N p_i \log p_i$$

We then have the *average information per symbol observed* as

$$I/N = -\sum_{i=1}^{n} p_i \log p_i$$

Note that $\lim_{x\to 0} x \log(1/x) = 0$ so we can define $p_i \log(1/p_i) = 0$ when $p_i = 0$ (i.e. adding terms of zero probability does not change the average information).

Definition (Entropy). Let X be a discrete random variable with alphabet $\mathcal{X}$ and probability mass function $p(x) = \Pr(X = x), x \in \mathcal{X}$, then the Entropy $H(X)$ of random variable X is defined as

$$H(X) = -\sum_{x\in\mathcal{X}} p(x) \log p(x)$$

we may also write $H_b(p)$ for the entropy of X using base b.

Entropy

Some immediate **Lemmas.**

1. $H(X) \geq 0$

2. $H_b(X) = (\log_b a) H_a(X)$

3. $H(X) = E\{-\log p(X)\}$ (expected value of the surprise)

It is important to recognize that the definitions of *information* and *entropy* depend only on the probability mass function of the random variable. In general, it doesn't make sense to talk about the *information* or the *entropy* of a source without specifying the probability distribution.

Entropy

It can certainly happen that two different observers of the same data stream have different models of the source, and hence associate different probability distributions to the source. The two observers will then assign difference values to the *entropy* associated with the source.

This somewhat fits with our intuition: two people listening to the same (albeit a boring) lecture can get very different information from the lecture. For example, without the appropriate background, one person might not understand anything at all, and therefore have a probability model of a completely random source, and therefore get much more information than the listener who understands, and can anticipate much of what goes on, and assigns a non-equal probabilities to successive words......

Entropy

Definition (Joint Entropy). The joint entropy $H(X,Y)$ of a pair of discrete random variables (X,Y) with a joint probability distribution $p(x,y)$ is defined as

$$H(X,Y) = -\sum_{x,y} p(x,y) \log p(x,y)$$

Definition (Conditional Entropy). If $(X,Y) \sim p(x,y)$, then the Conditional Entropy $H(Y|X)$ is defined as

$$H(Y|X) = -\sum_{x,y} p(x,y) \log p(y|x)$$

Mutual Information

The *mutual information* is a measure of the amount of information that one random variable contains about another random variable. It gives the reduction in the uncertainty of one random variable due to the knowledge of the other.

Definition (Mutual Information). Consider two random variables X and Y with a joint probability mass function $p(x,y)$ and marginal probability mass functions $p(x)$ and $p(y)$, then the Mutual Information $I(X;Y)$ is defined as the relative entropy between the joint distribution and the product distribution $p(x)p(y)$

$$I(X;Y) = D(p(x,y)\|p(x)p(y)) = \sum_{x,y} p(x,y) \log \frac{p(x,y)}{p(x)p(y)}$$

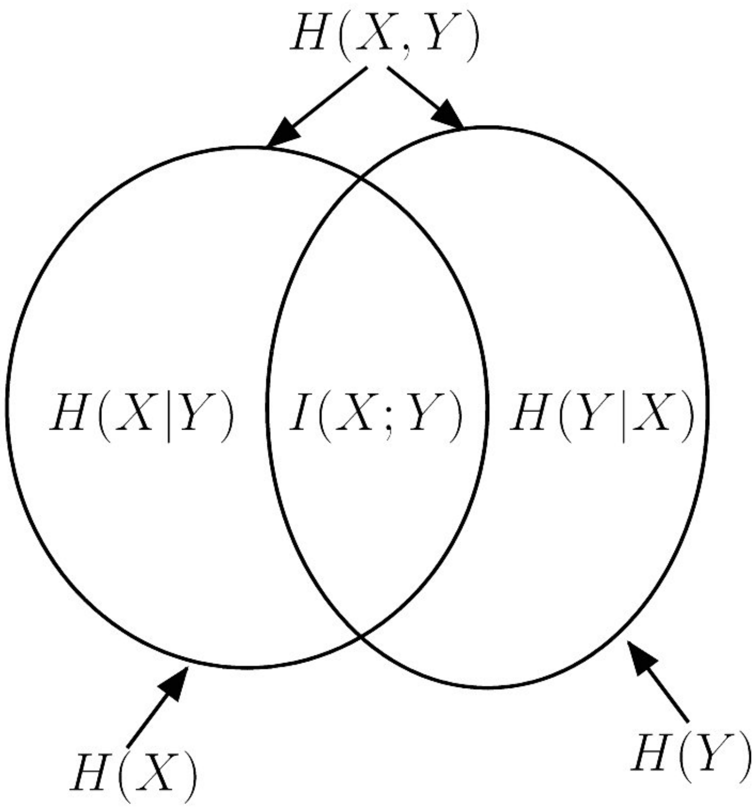
Mutual Information and Entropy

We can relate the mutual information of two random variables X and Y with the entropies as follows:

$$
\begin{aligned}
I(X;Y) &= H(X) - H(X|Y) \\
&= H(Y) - H(Y|X) \\
&= H(X) + H(Y) - H(X,Y) \\
&= I(Y;X) \\
I(X;X) &= H(X)
\end{aligned}
$$

Venn Diagram

We can visualize these relationships using a ‘Venn’ diagram



Channel Coding

If we are given a channel, we could ask what is the maximum possible information transfer through the channel? We would also like to know if it is possible to encode the data as to achieve the maximum.

These questions are answered in Shannon's channel coding theorem.

For any channel, there exists ways of encoding input symbols such that we can simultaneously utilize the channel as closely as we wish to the capacity, and at the same time have an error rate as close to zero as we wish.

Unfortunately, Shannon's proof is non-constructive, that is, it does not tell us how to construct the coding system to optimize the channel use, but only tells us that such a code exists.

Discrete Channels

A **discrete channel** is a system consisting of an input alphabet $\mathcal{X}$, an output alphabet $\mathcal{Y}$, and a probability transition matrix $p(y|x)$ that expresses the probability of observing the output symbol y given that we send the symbol x.

The channel is said to be **memoryless** if the probability distribution of the output depends only on the input at that time and is conditionally independent of previous channel inputs or outputs.

If the input depends on the past output, then we have a **channel with feedback**. Unless stated we will only consider channels without feedback.

Channel Capacity

The **information channel capacity** of a discrete memoryless channel is

$$C = \max_{p(x)} I(X;Y)$$

where the maximum is taken over all possible input distributions $p(x)$.

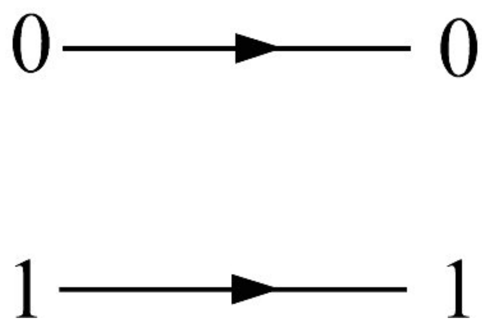
The **operational channel capacity** is the highest rate in bits per channel use at which information can be sent with arbitrarily low probability of error.

Shannon showed that information channel capacity $=$ operational channel capacity, therefore we just talk about *channel capacity*.

Examples of Channel Capacity

Noiseless Binary Channel
For the noiseless binary channel the binary input is reproduced exactly at the output. Any transmitted bits is therefore received without error; for each use of the channel, we can send 1 bit reliably to the receiver and the capacity is 1 bit.

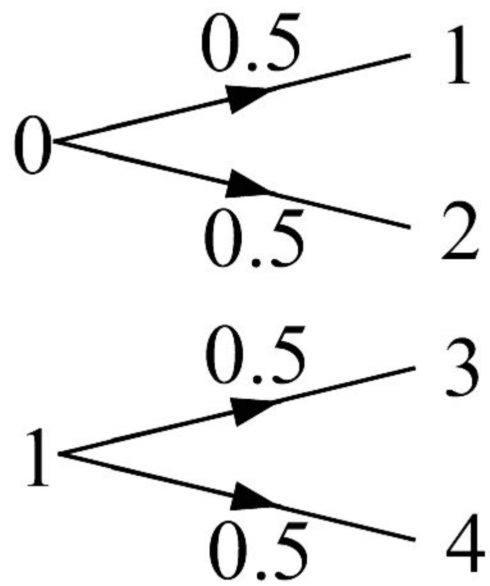
Can also calculate capacity $C = \max I(X;Y) = 1$ bit, achieved with $p(x) = (1/2, 1/2)$.



Noisy Channel with Nonoverlapping Outputs

The channel appears to be noisy, but really is not as the input can be determined from the output

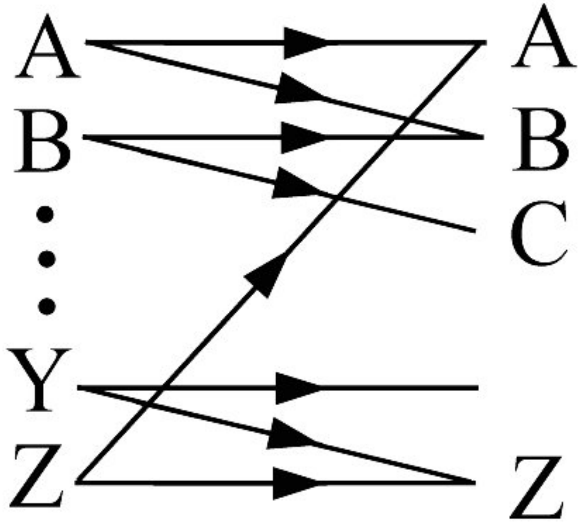
Hence, the channel is equivalent to the noiseless binary channel with a capacity of 1 bit. Again can get this from $C = \max I(X;Y) = 1$ bit, achieved with $p(x) = (1/2, 1/2)$.



Noisy Typewriter

The channel input is either received unchanged at the output with probability 0.5 or it is transformed into the next letter with probability 0.5. For an input of 26 symbols using every alternative input symbol we can transmit 13 symbols without error with each transmission, hence the capacity is $\log 13$ bits per transmission. Similarly, with uniform $p(x)$,

$$C = \max I(X;Y) = \max[H(Y) - H(Y|X)] = \max H(Y) - 1$$
$$= \log 26 - 1 = \log 13$$



Channel Codes

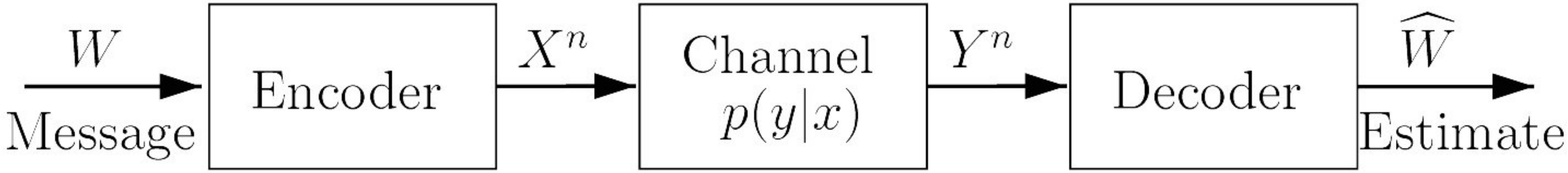
A (n, M) channel code consists of the following:

1. An index set $I = \{1, 2, \ldots, M\}$
2. An encoding function $X^n : I \to \mathcal{X}^n$
3. A decoding function $g : \mathcal{Y}^n \to I$

The conditional *probability of error* given index i was sent is

$$\lambda_i = \text{Pr}(\widehat{W} = i | W = i)$$

where W is the original index, and $\widehat{W} = g(Y^n)$ is the receiver guess of the transmitted index W.



The *maximal probability of error* $\lambda^{(n)}$ for a channel code is defined as

$$\lambda^{(n)} = \max_{i \in I} \lambda_i$$

The **rate** of an (n, M) code is

$$R = \frac{\log M}{n} \text{ bits per transmission}$$

A rate R is said to be **achievable** if there exists a sequence of $(n, 2^{nR})$ codes such that the maximal probability of error $\lambda^{(n)} \to 0$ as $n \to \infty$.

The **capacity** of a discrete memoryless channel is the supremum of all achievable codes.

The Channel Coding Theorem

Theorem 1. [Channel Coding Theorem] *All rates below capacity are achievable. Specifically, for every rate $R < C$, there exists a sequence of $(n, 2^{nR})$ codes with maximum probability probability of error $\lambda^{(n)} \to 0$. Conversely, any sequence of $(n, 2^{nR})$ codes with $\lambda^{(n)} \to 0$ must have $R \leq C$.*

Proof (sketch). We wish to show there exists a code and decoder having a small probability of error. Evaluating the probability of error of any system is not easy. Instead of constructing a good system and evaluating its error probability, we calculate the average probability of block error of *all* codes, and prove this average is small. There must then exist individual codes that have small probability of block error.

Continuous Channels

So far we have discussed only discrete channels; we shall now look at *continuous* channels.

If the discrete case is mastered, it is not difficult to handle the analogous definitions and results of the continuous case. For example, sums are replaced by integrals, etc. There are some important differences, however, and some care is needed. Calculations, for example, are generally more involved in the continuous case.

Entropy in the discrete case corresponds to *differential entropy* in the continuous case.

Differential Entropy

The **differential entropy** $h(X)$ of a continuous random variable X with probability density function $f(x)$ is defined as

$$h(X) = -\int_S f(x) \log f(x) dx$$

where S is the support set of the random variable (the set where $f(x) > 0$).

As in the discrete case, the differential entropy depends only on the probability function of the random variable.

Note that the differential entropy is defined only if the aforementioned integral and density function exist.

As mentioned above, by simply replacing the sums with integrals all the discrete channel definitions and theorems hold for probability density functions.

The Gaussian Channel

The most important continuous alphabet channel is the **Gaussian channel**. This a *time discrete* channel with, at time i, input X_i, noise Z_i which is drawn from a Gaussian distribution with variance N, and output Y_i , hence

$$Y_i = X_i + Z_i, \qquad Z_i \sim \mathcal{N}(0, N)$$

We must also have a power constraint on the input. We require for any codeword $(x_1, x_2, \ldots, x_n)$ transmitted over the channel that $E\{X^2\} \leq P$.

Capacity for Gaussian Channels

The **capacity** of the Gaussian channel with power constraint P is

$$C = \max_{p(x):E\{X^2\}\leq P} I(X;Y)$$

Expanding mutual information, we get

$$\begin{aligned} I(X;Y) &= h(Y) - h(Y|X) = h(Y) - h(X+Z|X) \\ &= h(Y) - h(Z|X) = h(Y) - h(Z) \end{aligned}$$

where Z is assumed independent of X.

The following bounds on entropy can be shown

$$h(Y) \leq \frac{1}{2}\log(2\pi e(P+N))$$
$$h(Z) = \frac{1}{2}\log(2\pi eN)$$

therefore the mutual information is given by

$$I(X;Y) = h(Y) - h(Z) \leq \frac{1}{2}\log(2\pi e(P+N)) - \frac{1}{2}\log(2\pi eN) = \frac{1}{2}\log(1+P/N)$$

Hence the information capacity of the Gaussian channel is

$$C = \max_{E\{X^2\}\leq P} I(X;Y) = \frac{1}{2}\log(1+P/N)$$

with the maximum obtained when $X \sim \mathcal{N}(0,P)$.

Outline

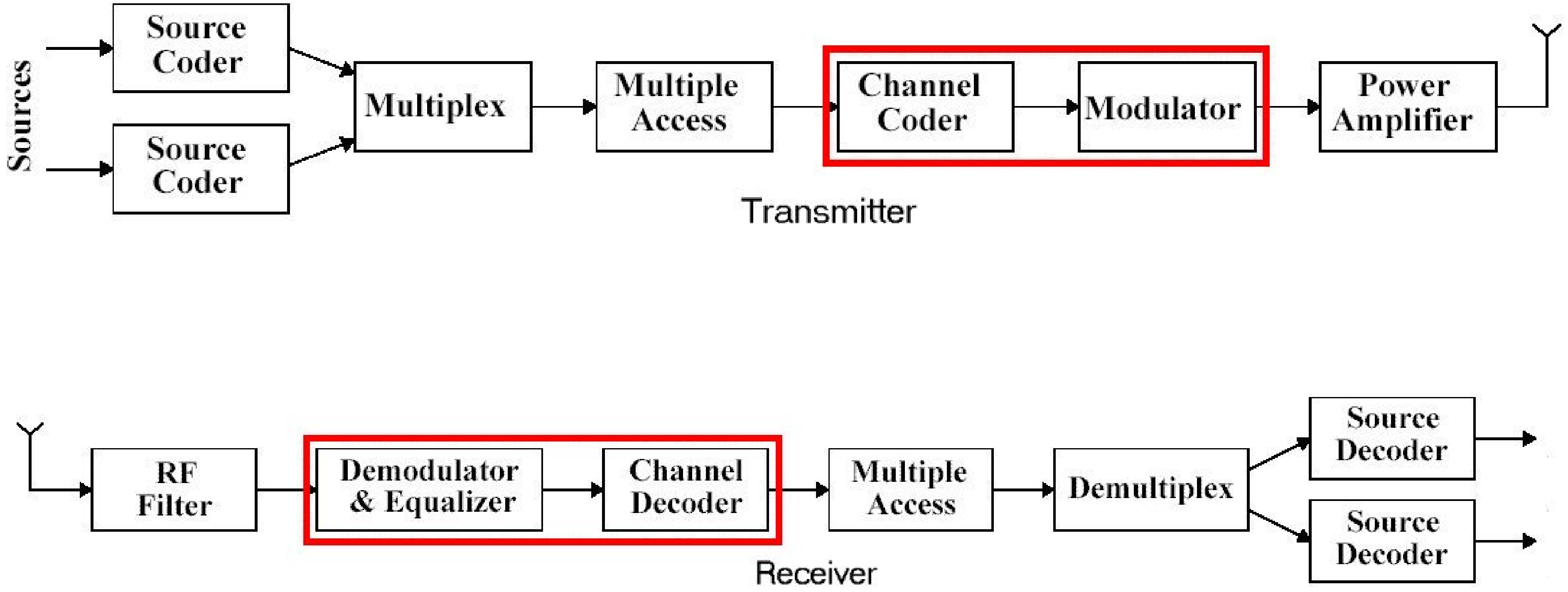
Wireless Channels :

- Wired vs Wireless
- Channel Characteristics and Modelling
- Channel Capacity

Communications Systems :

- Channel Coding and Decoding
- Modulation and Demodulation
- Equalization

Communication Systems



Outline

Wireless Channels :

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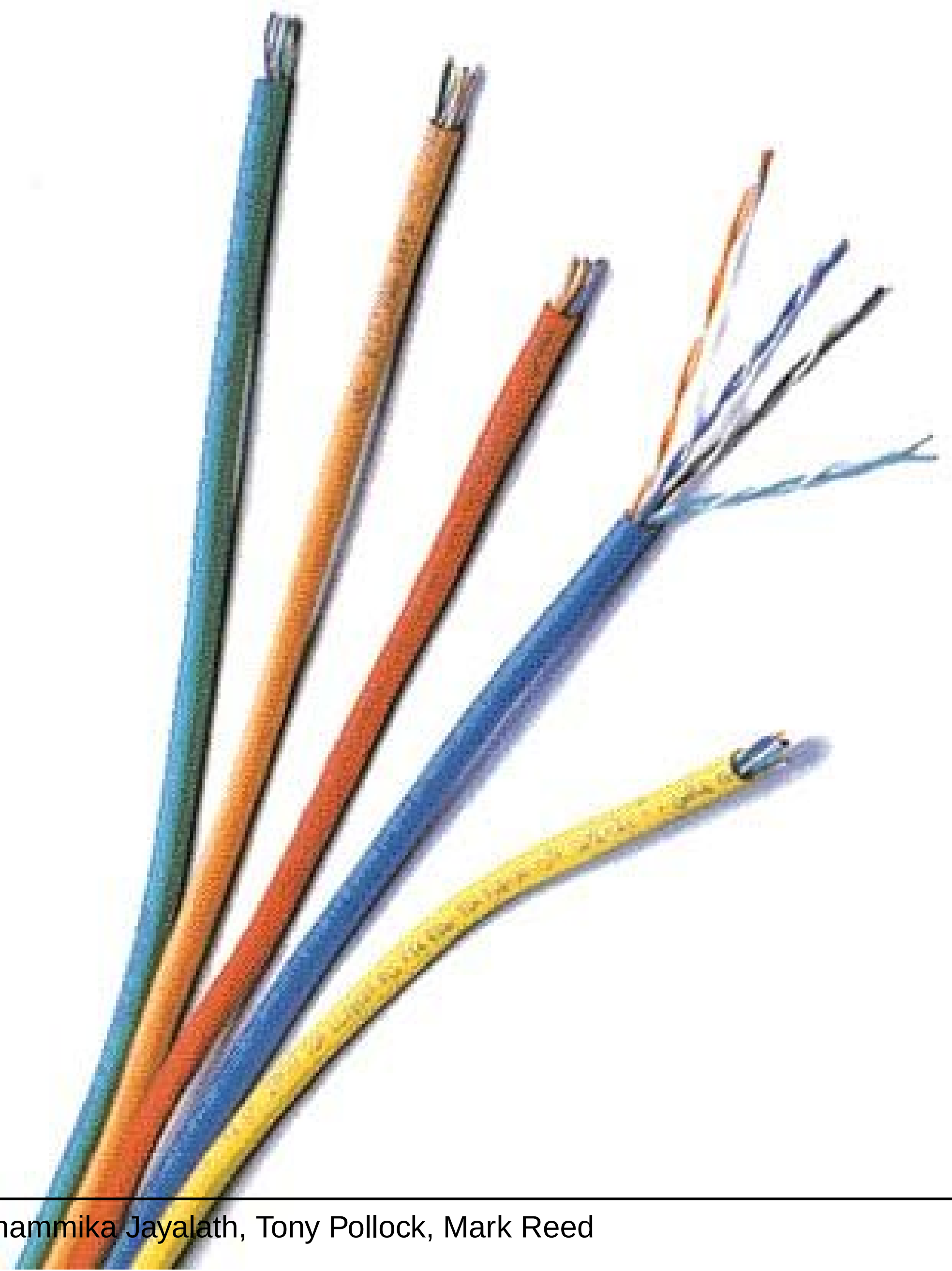
Communications Systems

Channel Coding and Decoding :

- Error Control Coding
- Block Codes
- Convolutional Codes
- ML Decoding



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Error Control Coding

Error control for data integrity can be exercised by **forward error control** (FEC).

Channel Encoder accepts message bits and adds redundancy according to some rule, and produces encoded data at higher bit rate.

Channel Decoder exploits the redundancy to decide which messages bits were most likely sent.

Goal is to minimize the effect of channel noise (minimize the number of received bits in error).

Trade-off: FEC adds **system complexity** and requires more **transmission bandwidth**.

Note, error detection only codes are also used, here the receiver uses an *automatic-repeat-request* (ARQ) for detected errors.

Error Control Coding

Historically codes are classified whether they have memory or not in the encoders:

Block Codes: (n, k) block code uses an encoder which accepts a k-bit message block and adds $n - k$ redundant (parity) bits (algebraically related to the k message bits) to produce a n-bit codeword. The code has a rate $r = k/n$.

Convolutional Codes: the encoding operation can be viewed as a *time convolution* of the input sequence with the impulse response of the encoder. The duration of the impulse response equals the memory of the encoder. The encoder accepts message bits as a continuous sequence and generates a continuous sequence of encoded bits at a higher bit rate.

Block Codes - Properties

Linear: any two code words in the code can be added in modulo-2 arithmetic to produce a third code word in the code.

Code Distance: the number of elements in which two codevectors $\mathbf{c}_i$ and $\mathbf{c}_j$ differ

$$d(\mathbf{c}_i, \mathbf{c}_j) = \sum_{\ell=1}^{n} \mathbf{c}_i(\ell) \oplus \mathbf{c}_j(\ell) \ (\text{modulo-}q)$$

For binary codes ($q = 2$) this is the *Hamming Distance*.

Minimum Distance: smallest code distance between any pair of codevectors.

Code Weight: the number of nonzero elements in the codeword.

Systematic: the parity bits are appended to the end of the information bits.

Block Codes - Generation

Consider $(6, 3)$ binary code ($c_i \in$ GF(2)) with structure $c = c_1\, c_2\, c_3\, c_4\, c_5\, c_6$ and parity code constraints

$$c_4 = c_1 \oplus c_2$$

$$c_5 = c_2 \oplus c_3$$

$$c_6 = c_1 \oplus c_2 \oplus c_3$$

which can generate the (systematic) codeword

$$[c_1\, c_2\, c_3\, c_4\, c_5\, c_6] = [c_1\, c_2\, c_3] \begin{bmatrix} 1 & 0 & 0 & 1 & 0 & 1 \\ 0 & 1 & 0 & 1 & 1 & 1 \\ 0 & 0 & 1 & 0 & 1 & 1 \end{bmatrix}$$

i.e., codeword vector $\mathbf{c}$ generated from message vector $\mathbf{m}$ is given by

$$\mathbf{c} = \mathbf{mG}$$

using **generator matrix** $\mathbf{G}$.

Block Codes - Parity Check

The codeword c is constrained by the parity check equations

$$c_1 \oplus c_2 \oplus c_4 = 0$$

$$c_2 \oplus c_3 \oplus c_5 = 0$$

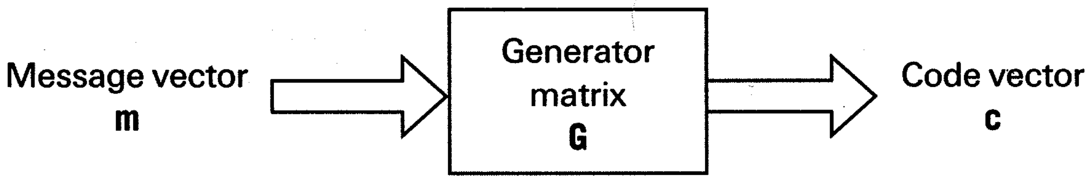
$$c_1 \oplus c_2 \oplus c_3 \oplus c_6 = 0$$

or in matrix form

$$\begin{bmatrix} 1 & 1 & 0 & 1 & 0 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 \\ 1 & 1 & 1 & 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} c_1 \\ c_2 \\ c_3 \\ c_4 \\ c_5 \\ c_6 \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ 0 \end{bmatrix}$$

$$\mathbf{H}\mathbf{c}^T = \mathbf{0}$$

Block Codes - Parity Check



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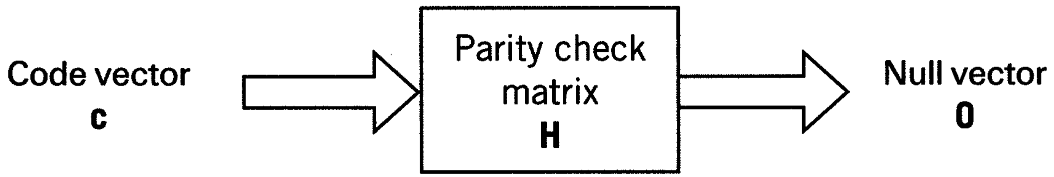


(b)

Block Codes - Error Detection

A received vector $\mathbf{r}$ due to transmitted codeword $\mathbf{c}$ is a valid codeword if $\mathbf{H}\mathbf{r}^T = \mathbf{0}$. If $\mathbf{r}$ is not a valid codeword

$$\mathbf{r} = \mathbf{c} + \mathbf{e}$$

where $\mathbf{e}$ is the **error pattern**, with

$$e_i = \begin{cases} 1 & \text{if an error has occurred in the } i\text{th location} \\ 0 & \text{otherwise} \end{cases}$$

Syndrome Decoding uses the syndrome $\mathbf{s} = \mathbf{r}\mathbf{H}^T$ to determine the most probable error $\mathbf{e}$ to enable error correction.

Note that the syndrome depends only on the error pattern (and not the transmitted codeword), and all error patterns that differ by a codeword have the same syndrome.

Block Codes - Error Correction

A (n, k) linear block code has the power to correct all error patterns of weight t or less if, and only if,

$$d(\mathbf{c}_i, \mathbf{c}_j) \geq 2t + 1 \text{ for all } \mathbf{c}_i \text{ and } \mathbf{c}_j$$

By definition the smallest distance between any pair of code vectors is the minimum distance of the code, d_{min}. Therefore,

A (n, k) linear block code of minimum distance d_{min} can correct up to t errors if, and only if,

$$t \leq \lfloor (d_{\text{min}} - 1)/2 \rfloor$$

The minimum distance of the code is an important parameter of the code and determines its error-correcting capability.

Convolutional Codes

Block codes require buffers to store the entire message block before generating the associated codeword.

Convolutional coding operates on the incoming message sequence **continuously** in a serial manner.

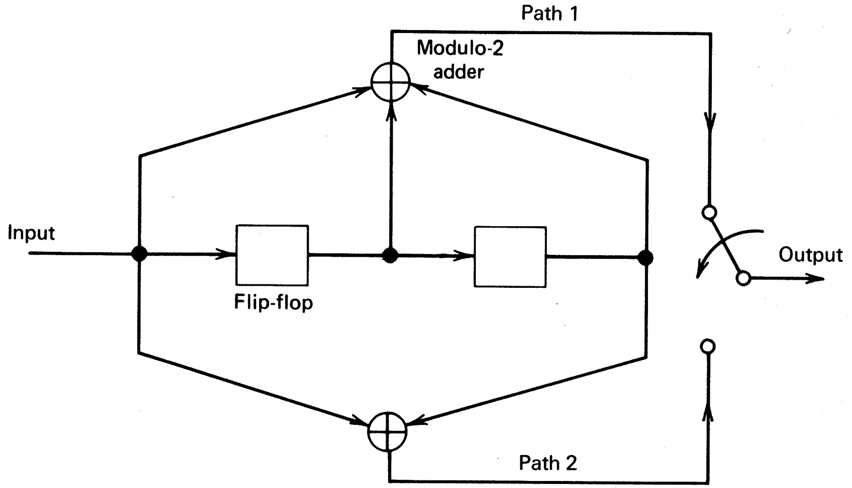
For an encoder of memory length M an L-bit message sequence produces a codeword of length $n(L+M)$ bits, giving code rate

$$r = \frac{L}{n(L+M)}$$

Typically $L \gg M$, hence $r \simeq 1/n$.

Constraint Length: number of shifts over which a single message bit can influence the encoder output, $K = M + 1$.

Convolutional Codes - Generation



Example: rate 1/2, constraint length 3 encoder

Each path connecting the output to the input of a convolutional encoder may be characterized in terms of its **impulse response**, defined as the response of the path to a symbol 1 applied to the input, with each flip-flop in the encoder set initially in the zero state.

Generator Polynomial: the unit delay transform of the impulse response for each path:

$$g^{(1)}(D) = 1 + D + D^2 \qquad g^{(2)}(D) = 1 + D^2$$

Convolutional Codes - Generation

The output sequence (codeword) is the time convolution of the message and impulse response of encoder.

For message sequence in polynomial form $m(D)$, time convolution is simply **multiplication** in the D domain (Fourier Transform):

$$c(D) = g(D)m(D)$$

Example: Consider message sequence (10011), we have polynomial representation $m(D) = 1 + D^3 + D^4$

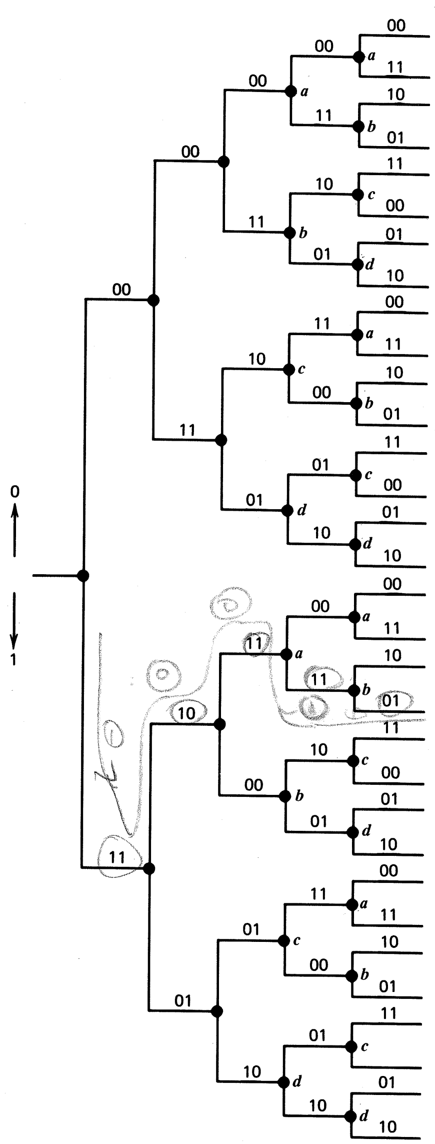
Path 1 output: $c^{(1)}(D) = g^{(1)}(D)m(D) = 1 + D + D^2 + D^3 + D^6$

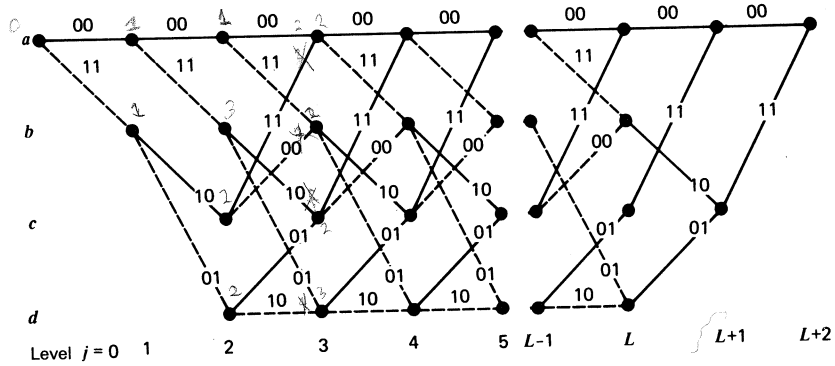
Path 2 output: $c^{(2)}(D) = g^{(2)}(D)m(D) = 1 + D^2 + D^3 + D^4 + D^5 + D^6$

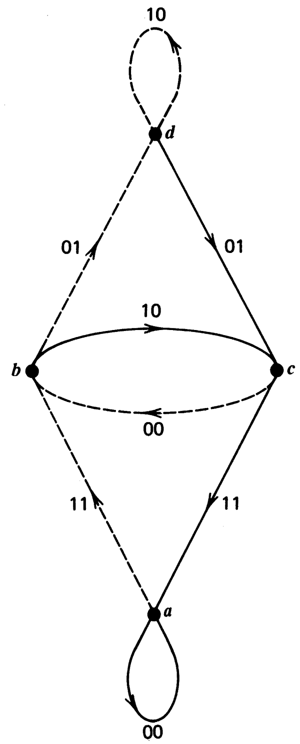
Paths 1 (1111001) and 2 (1011111) are then multiplexed

$$\mathbf{c} = (11, 10, 11, 11, 01, 01, 11)$$

Convolutional Codes - Code Tree, Trellis and State







Convolutional Codes - Maximum Likelihood Decoding

$\mathbf{m}$	message vector
$\mathbf{c}$	code vector
$\mathbf{r}$	received vector - differs from transmitted code vector due to channel noise

Given received vector $\mathbf{r}$, decoder is required to make an estimate $\hat{\mathbf{m}}$ of message vector.

Since one-to-one correspondence between $\mathbf{m}$ and $\mathbf{c}$ decoder may equivalently produce an estimate of code vector $\hat{\mathbf{c}}$.

Maximum Likelihood Decoder:

Choose the estimate $\hat{\mathbf{c}}$ *for which the*
log-likelihood function $\ln p(\mathbf{r}|\mathbf{c})$ *is maximum.*

ML gives the optimal decoding for equiprobable messages, i.e. the *probability of decoding error* is minimized.

Convolutional Codes - Viterbi Algorithm

Can decode a convolutional code by choosing a path in the code tree/trellis whose coded sequence differs from a received sequence in the fewest number of places.

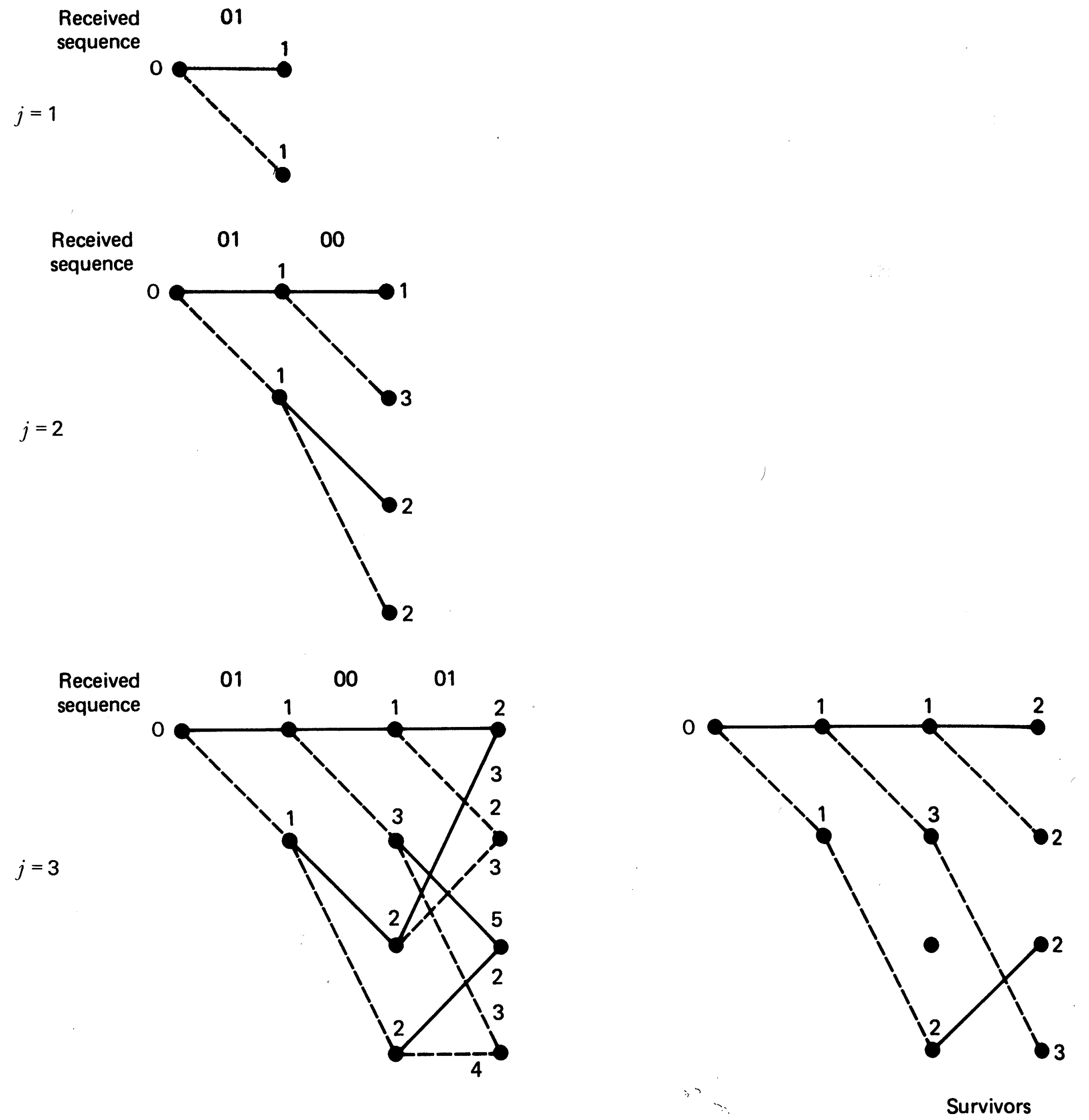
The Viterbi algorithm computes a **metric** for every possible path in the trellis.

The metric for a particular path is is defined as the **Hamming distance** between the coded sequence represented by that path and the received sequence.

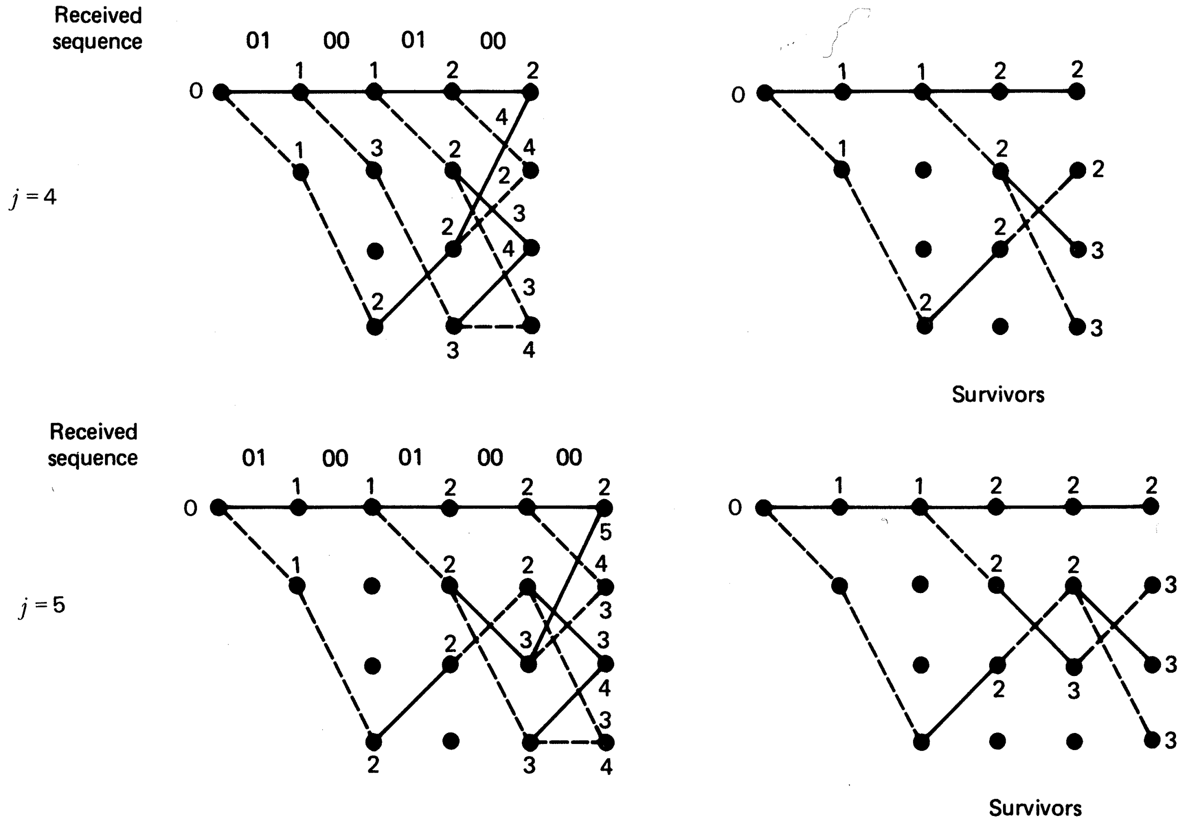
For each state of in the trellis the algorithm compares the two paths entering it, the path with the lowest metric is retained - the **survivor path**. This is repeated for every level j of the trellis.

Example: suppose encoder generates an all-zero sequence sent over a binary symmetric channel giving received sequence $(0100010000\ldots)$

Convolutional Codes - Viterbi Algorithm



Convolutional Codes - Viterbi Algorithm



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Wireless Channels :

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Communications Systems :

- Channel Coding and Decoding
- **Modulation and Demodulation**
- Equalization

Communications Systems

Modulation and Demodulation :

- Modulation
- Amplitude Modulation
- Angle Modulation
- Coherent Binary Phase Shift Keying (BPSK)
- Coherent Binary Frequency Shift Keying (BFSK)
- Coherent Quadri-Phase Shift Keying (QPSK)
- M-ary Modulation Techniques

Modulation/Demodulation

A communication system is required to transmit information-bearing signals or baseband signals through a communication channel separating the transmitter from the receiver.

Baseband designates the band of frequencies representing the original signal (source).

To proper utilize the channel we generally require a shift of the range of baseband frequencies into other frequency ranges suitable for transmission, and a corresponding shift back to baseband frequencies after reception. Consider an audio signal, to send this over any significant distances a shift in the range of frequencies is required for the system to operate satisfactorily.

Modulation: the process by which some characteristic of a *carrier wave* is varied in accordance with a *modulating wave (signal)*.

Demodulation: restoring the original baseband signal after reception.

Amplitude Modulation (AM)

Amplitude Modulation (AM) is the process in which the amplitude of the carrier wave is varied about a mean value, linearly with the (continuous) baseband message signal.

Consider a sinusoidal carrier wave $c(t)$

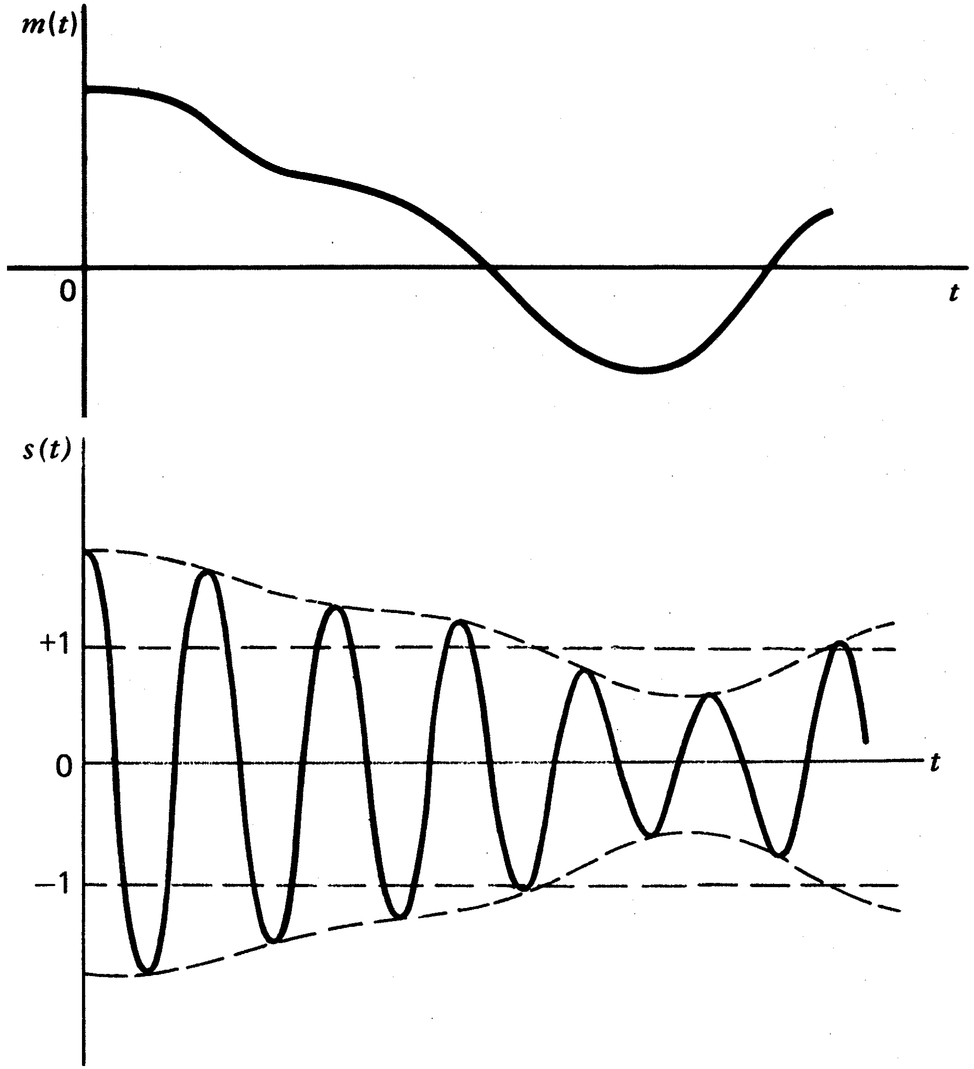
$$c(t) = A_c \cos(2\pi f_c t)$$

with *carrier amplitude* A_c and *carrier frequency* f_c, then the modulated baseband message signal $m(t)$ is

$$s(t) = A_c[1 + k_a m(t)] \cos(2\pi f_c t)$$

k_a is the amplitude sensitivity constant of the modulator.

AM



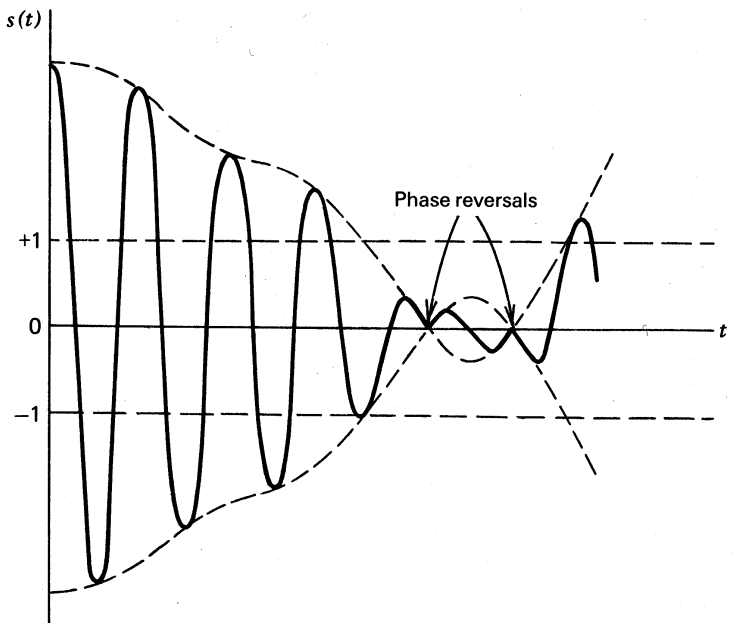
AM

The envelope of the modulated signal $s(t)$ has essentially the same shape as the baseband signal provided:

1. The amplitude of $k_a m(t)$ is always less than unity

$$|k_a m(t)| < 1 \quad \text{for all } t$$

If $|k_a m(t)| > 1$ the carrier wave becomes **overmodulated**, resulting in carrier phase reversals.

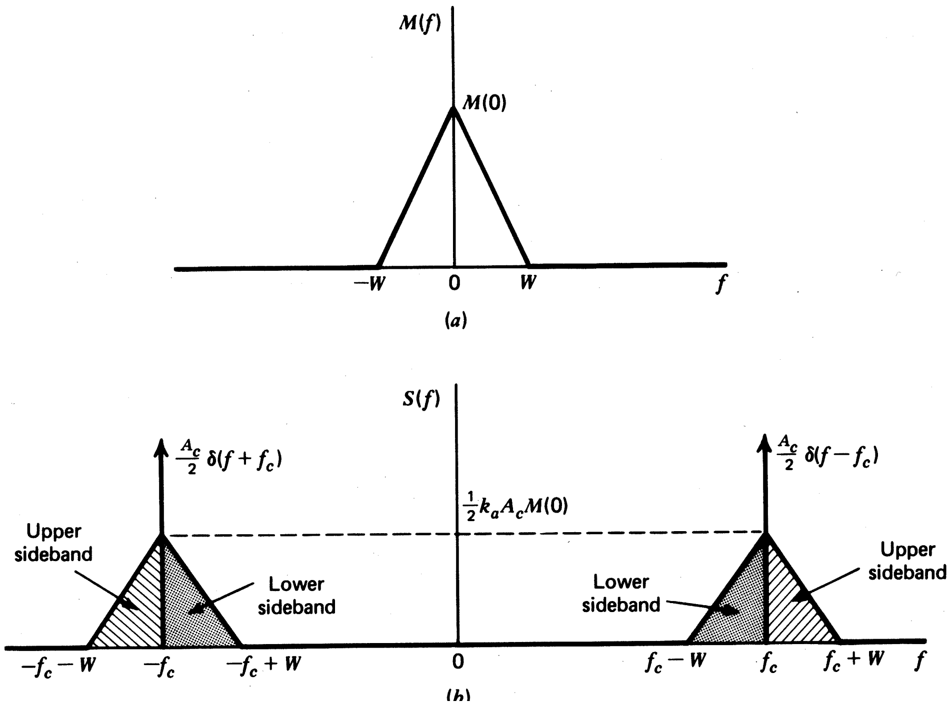


AM

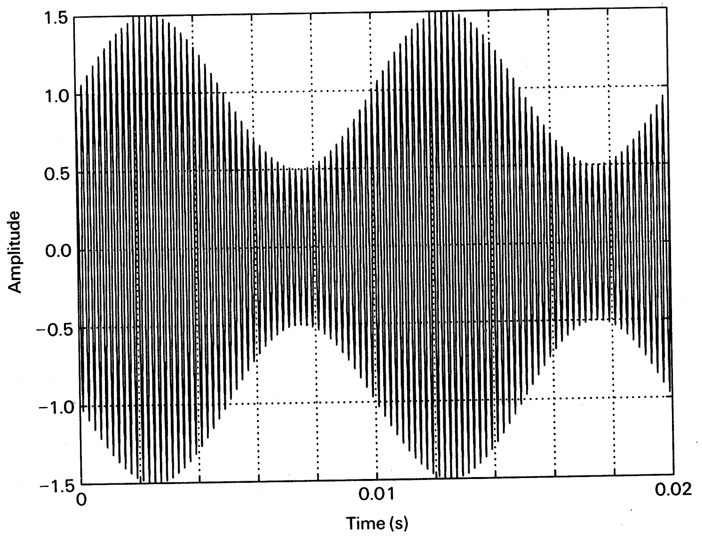
2. The carrier frequency is much greater than the highest frequency component of the message signal (**message bandwidth**).

$$f_c \gg W$$

If this is not satisfied the envelope cannot be visualized (and therefore detected) satisfactorily.



AM Envelope Detection



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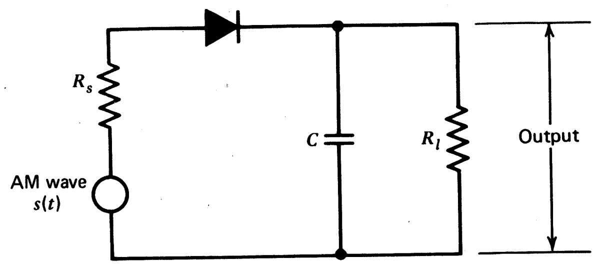

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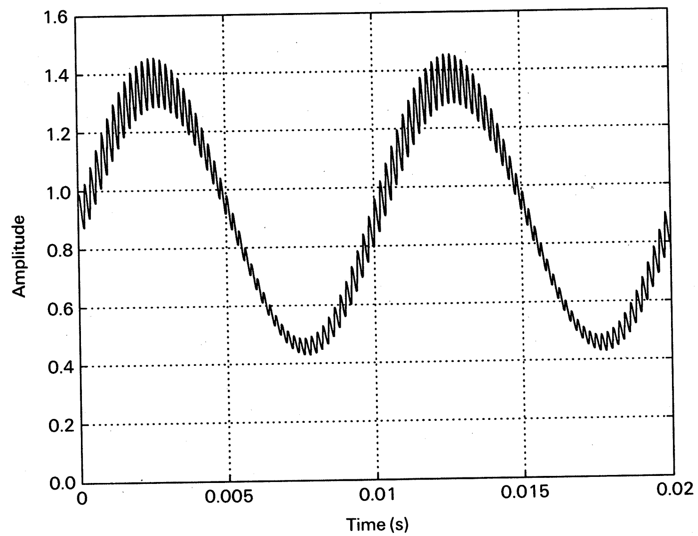


AM Virtues and Limitations

Amplitude modulation is the oldest modulation method. It is easy to generate and reverse, is cheap to construct, but

AM is wasteful of power - the carrier wave is completely independent of the baseband signal - transmission of it is therefore a waste of power.

AM is wasteful of bandwidth - upper and lower sidebands are uniquely related by virtue of their symmetry about the carrier - knowledge of one sideband is enough to determine the other.

These issues can be overcome (however there is now an increase in complexity):

Double sideband-suppressed carrier (DSB-SC) modulation

Vestigal sideband (VSB) modulation

Single sideband (SSB) modulation

Angle Modulation

Angle modulation is where the angle of the carrier wave is varied according to the baseband signal. Here the amplitude of the carrier remains constant.

Angle modulation provides better discrimination against noise and interference than amplitude modulation (however it requires larger transmission bandwidths).

Let $\theta_i(t)$ denote the *angle* of a modulated sinusoidal carrier (and is a function of message signal), then the **angle-modulated wave** is

$$s(t) = A_c \cos[\theta_i(t)]$$

There are an infinite number of ways to vary the angle $\theta_i(t)$ with the message (baseband) signal. Here we consider two commonly used methods, phase modulation and frequency modulation.

Angle Modulation - Phase Modulation (PM)

Angle modulation in which the angle $\theta_i(t)$ is varied linearly with the message signal $m(t)$

$$\theta_i(t) = 2\pi f_c t + k_p m(t)$$

$2\pi f_c t$ represents the angle of the unmodulated carrier.

k_p is the phase sensitivity of the modulator.

The phase-modulated signal is then

$$s(t) = A_c \cos[2\pi f_c t + k_p m(t)]$$

Angle Modulation - Frequency Modulation (FM)

Angle modulation in which the instantaneous frequency $f_t(t)$ is varied linearly with the message signal $m(t)$

$$f_i(t) = f_c + k_f m(t)$$

f_c is the frequency of the unmodulated carrier.

k_f is the frequency selectivity of the modulator.

Integrating wrt time gives

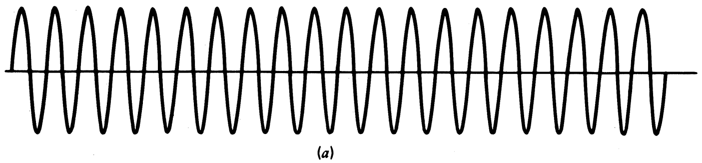
$$\theta_i(t) = 2\pi f_c t + 2\pi k_f \int_0^t m(t)dt$$

The frequency-modulated signal is then

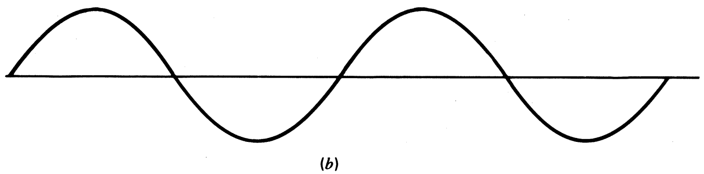
$$s(t) = A_c \cos[2\pi f_c t + 2\pi k_f \int_0^t m(t)dt]$$

Continuous Wave Modulation

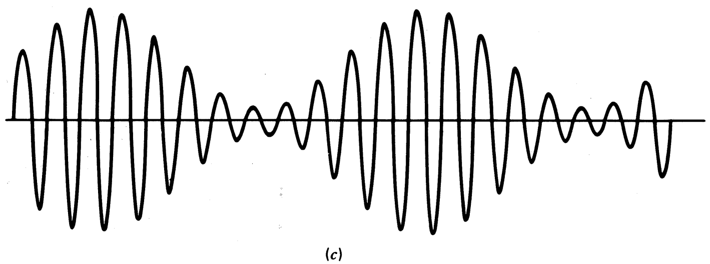
Carrier wave



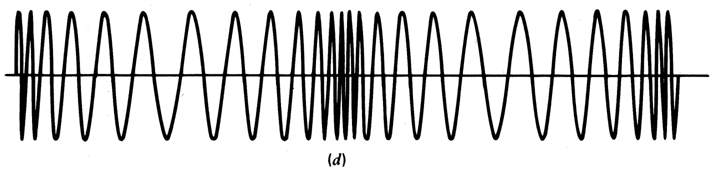
modulating signal



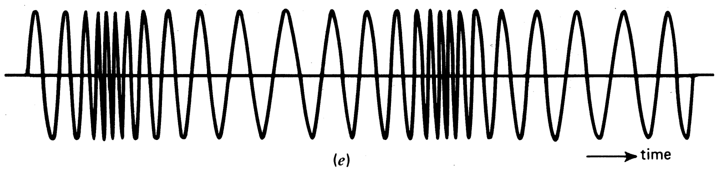
AM



PM



FM



Digital Modulation

Amplitude and angle modulation techniques varied a parameter of the sinusoidal carrier wave in accordance with a continuous wave message signal.

In **digital modulation** the message information is a discrete data stream which is then modulated onto a carrier (usually sinusoidal) with fixed frequency limits imposed by a band-pass channel of interest.

The modulation process involves **switching (keying)** the *amplitude, frequency*, or *phase* of a sinusoidal carrier in some fashion in accordance with the incoming data.

Three basic signalling schemes

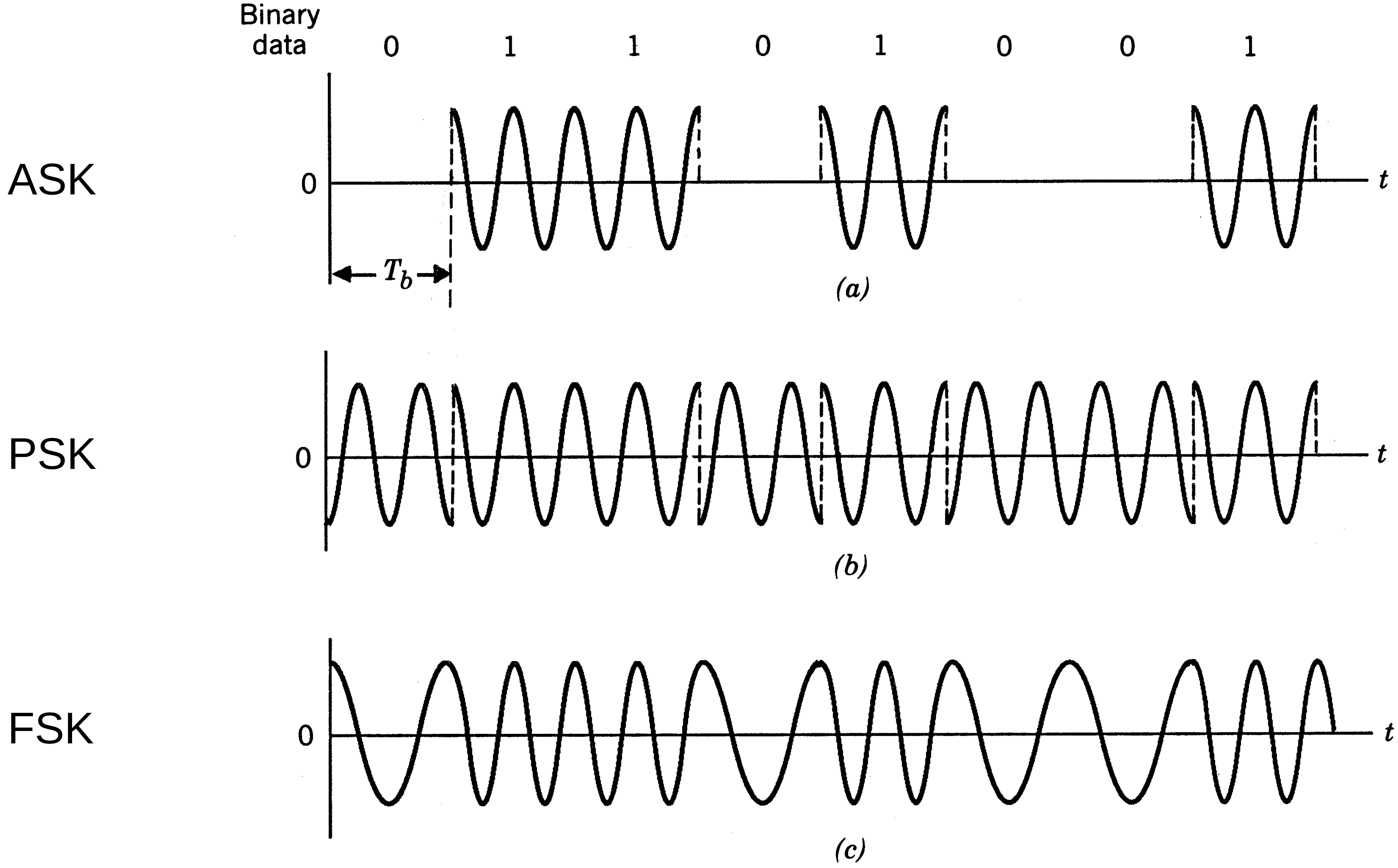
amplitude-shift keying (ASK)

frequency-shift keying (FSK)

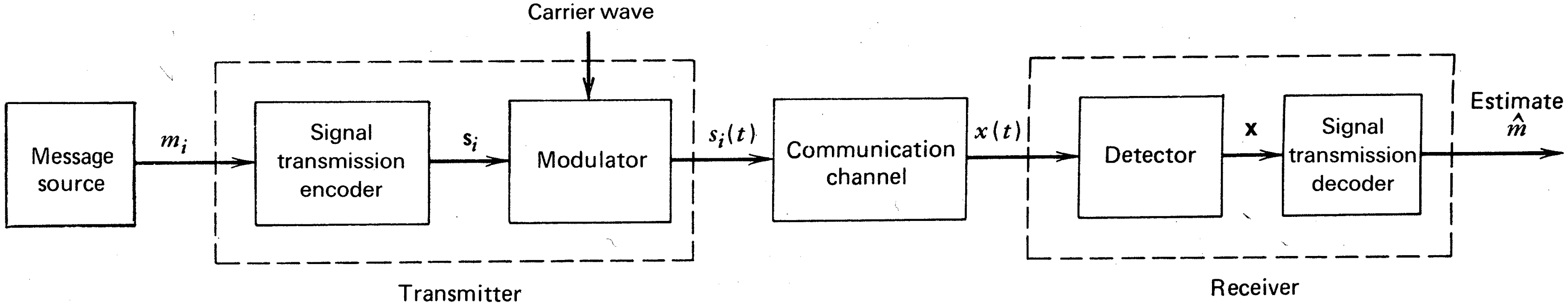
phase-shift keying (PSK)

These may be viewed as special cases of AM, FM, and PM.

Digital Modulation



Digital Modulation



Message Source: emits one *symbol* from an alphabet of M symbols every T seconds.
Encoder: produces N element vector $\mathbf{s}_i$ for each message symbol.
Modulator: constructs a *distinct* signal $s_i(t)$ of duration T seconds as the representation of the symbol m_i.

Detector: observes $x(t)$ for T seconds and produces observation vector $\mathbf{x}$.
Decoder: uses knowledge of modulation format and source distribution to produce estimate $\hat{m}$.

Digital Modulation - Signal Representation

Represent the signal as an expansion of orthonormal basis functions

$$s_i(t) = \sum_{j=1}^{N} s_{ij}\phi_j(t)$$

with coefficients

$$s_{ij} = \int_0^T s_i(t)\phi_j(t)dt$$

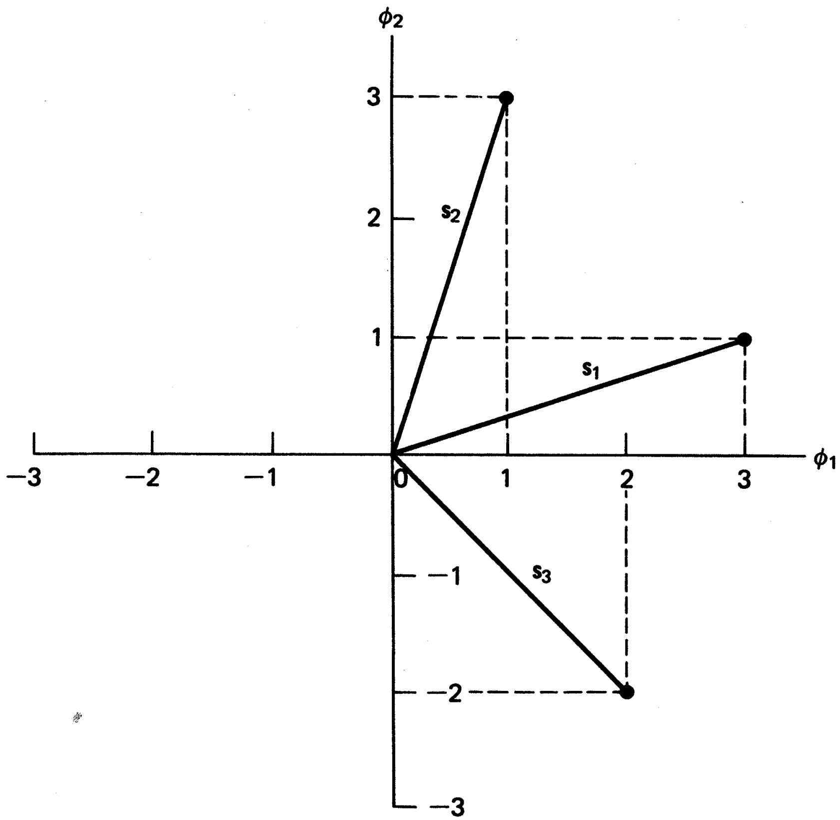
Hence, each signal in the set $\{s_i(t)\}$, $i = 1, 2, \ldots, M$ is completely determined by the vector of coefficents

$$\mathbf{s}_i = \begin{bmatrix} s_{i1} \\ s_{i2} \\ \vdots \\ s_{iN} \end{bmatrix}$$

Digital Modulation - Signal Representation

Signal space: an N-dimensional Euclidean space, with N mutually perpendicular axes $\{\phi_1, \phi_2, \ldots, \phi_N\}$, in which we may visualize the set of signal vectors $\{\mathbf{s}_i | i = 1, 2, \ldots, M\}$ as defining a set of points.

$N = 2$, $M = 3$



Digital Modulation - Detection

The task of the receiver is to minimize the *average probability of symbol error*

$$P_e = \sum_{i=1}^{M} P(\hat{m} \neq m_i) P(m_i)$$

We will always assume **time synchronized** transmitter and receiver (the receiver knows the instants of time when the modulation changes state).

There are two cases of **phase synchronization** between transmitter and receiver:

Coherent Detection: receiver is phase locked to transmitter

Noncoherent Detection: no phase synchronism between transmitter and receiver.

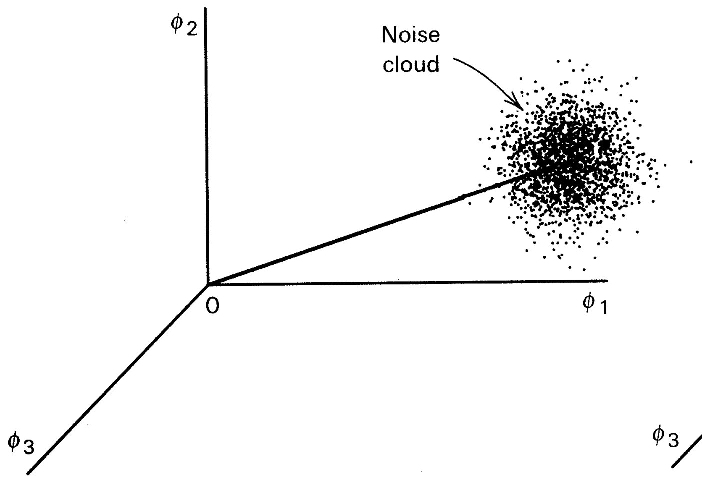
Digital Modulation - Coherent Detection in Noise

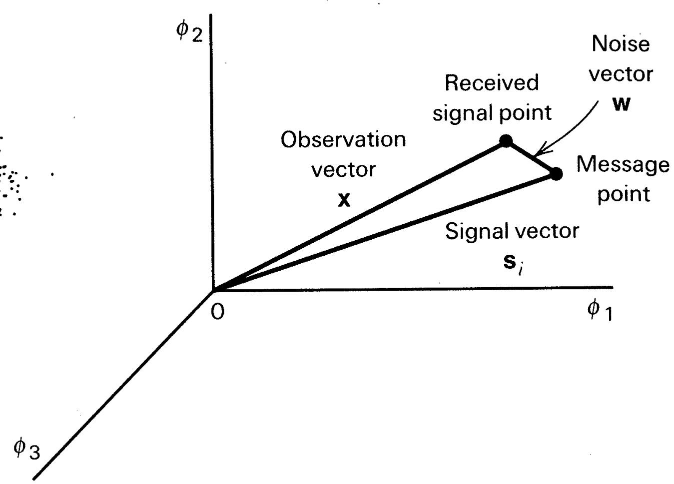
Consider the signal received in the presence of white Gaussian noise $w(t)$

$$x(t) = s_i(t) + w(t)$$

When the received signal is passed through a bank of correlators we have

$$\mathbf{x} = \mathbf{s}_i + \mathbf{w}$$



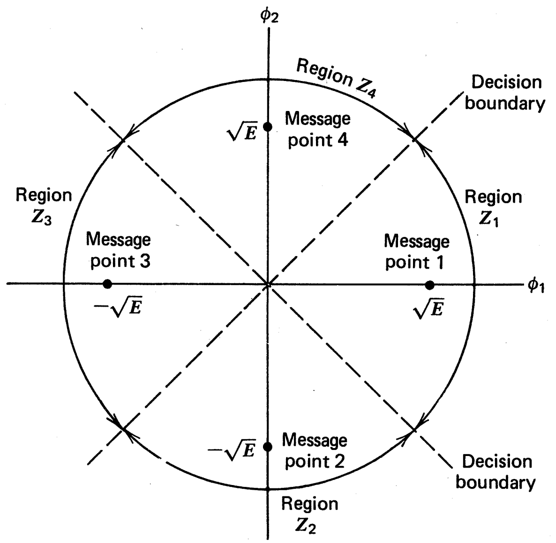


Digital Modulation - Coherent Detection in Noise

Given observation vector $\mathbf{x}$ *we have to perform a mapping from* $\mathbf{x}$ *to an estimate* $\hat{m}$ *of the transmitted symbol,* m_i*, in a way that would minimise the probability of an error in the decision making process.*

Maximum Likelihood Decision observation vector $\mathbf{x}$ lies in region Z_i if the Euclidean distance $\| \mathbf{x} - \mathbf{s}_k \|$ is minimum for $k = i$.

Maximum likelihood decision rule is simply to choose the message point closest to the received signal point.

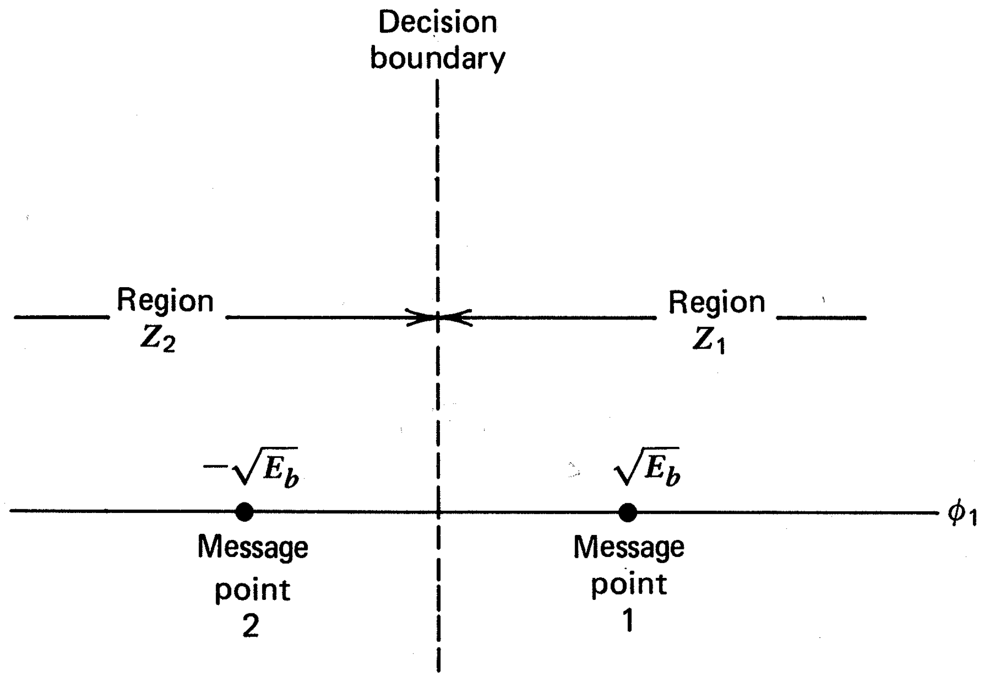


Coherent Binary PSK (BPSK)

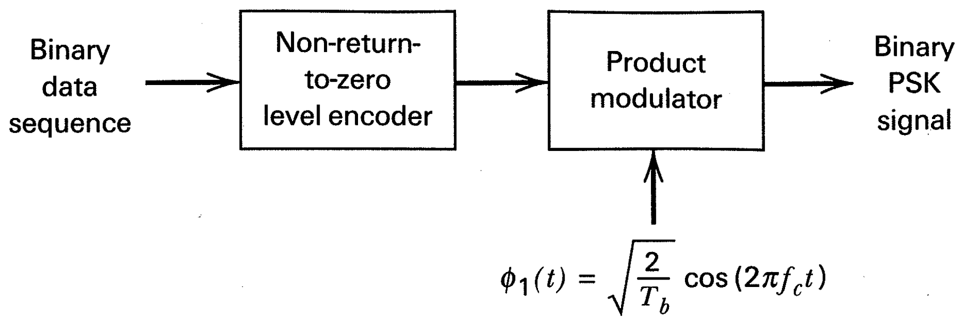
$$s_1(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t)$$

$$s_2(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t + \pi) = -\sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t)$$

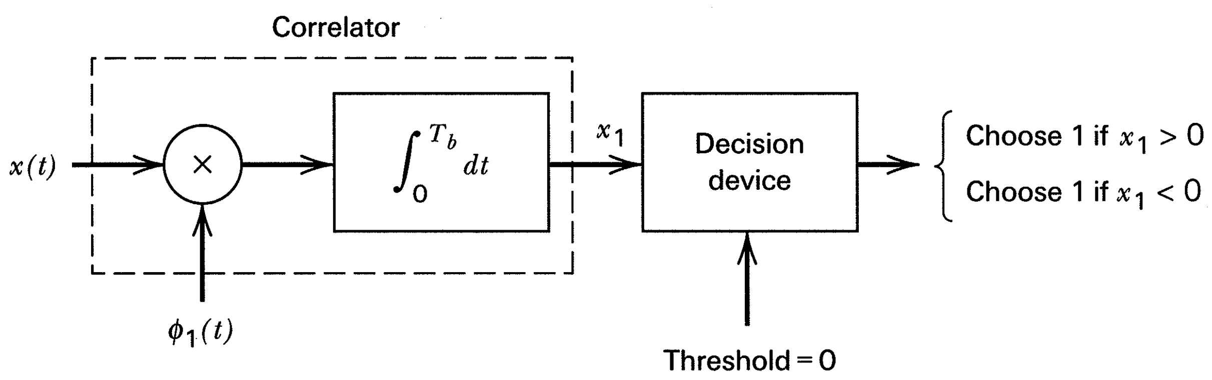
$$\phi_1(t) = \sqrt{\frac{2}{T_b}}\cos(2\pi f_c t), \qquad 0 \le t < T_b$$



BPSK - transmitter and receiver



(a)



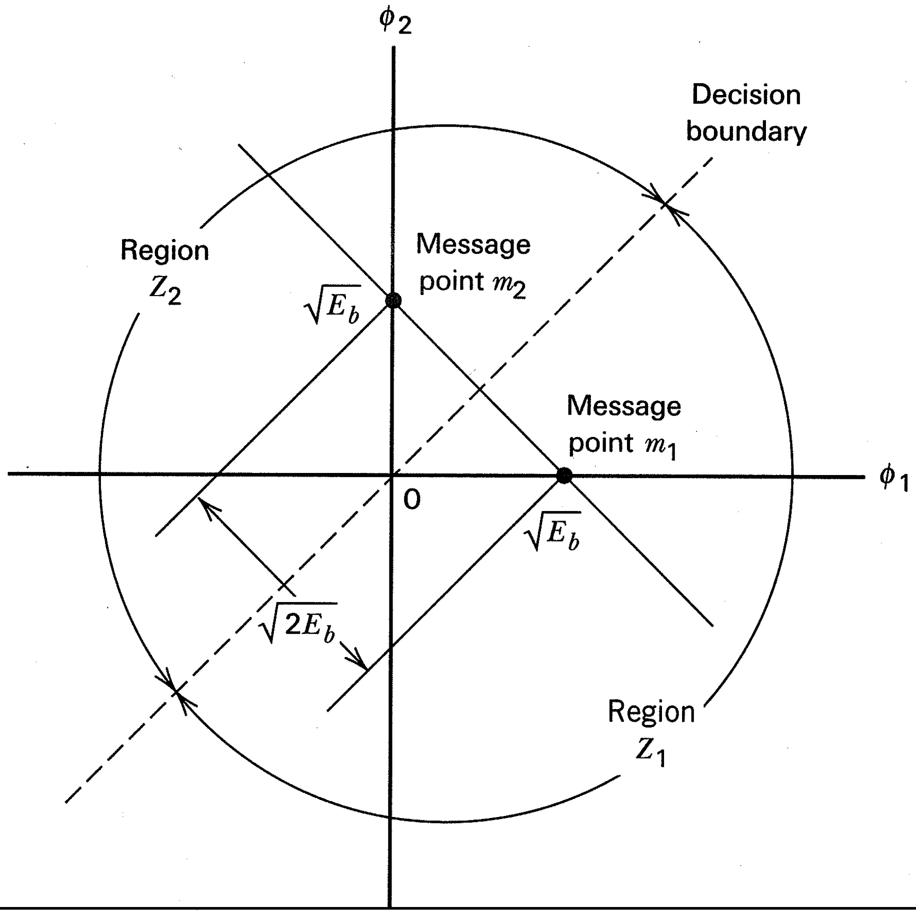
(b)

Coherent Binary FSK (BFSK)

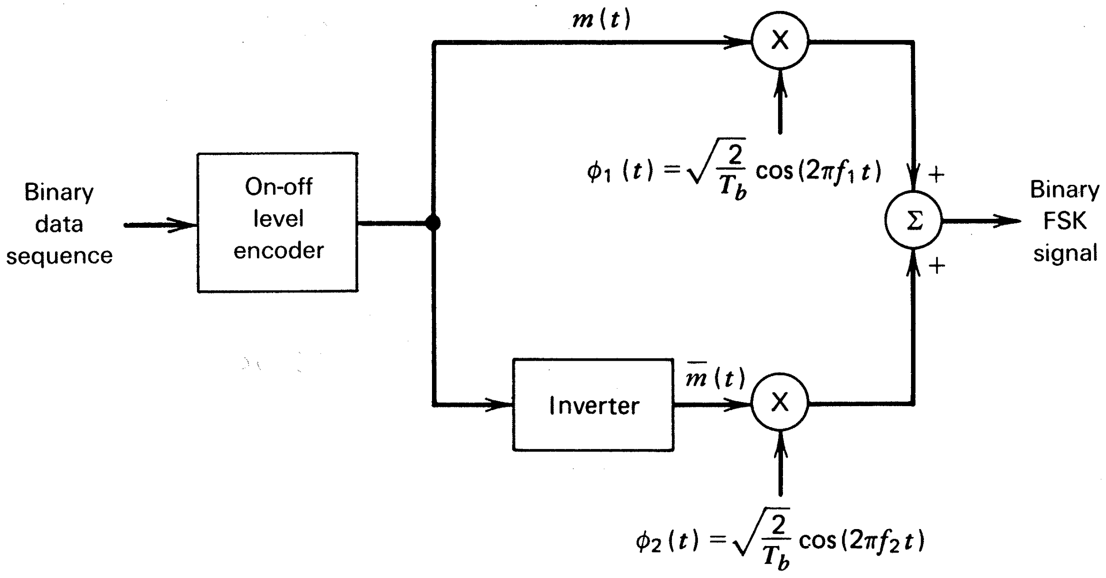
$$s_i(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_i t), \qquad f_i = \frac{n_c + i}{T_b}, \quad n_c \in \mathbb{Z}$$

$$\mathbf{s}_1 = \begin{bmatrix} \sqrt{E_b} \\ 0 \end{bmatrix}$$

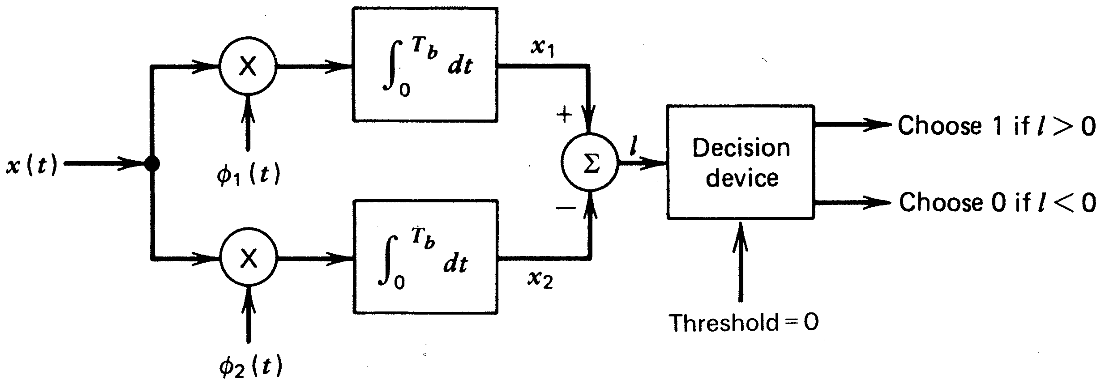
$$\mathbf{s}_2 = \begin{bmatrix} 0 \\ \sqrt{E_b} \end{bmatrix}$$



Coherent Binary FSK (BFSK)



(a)

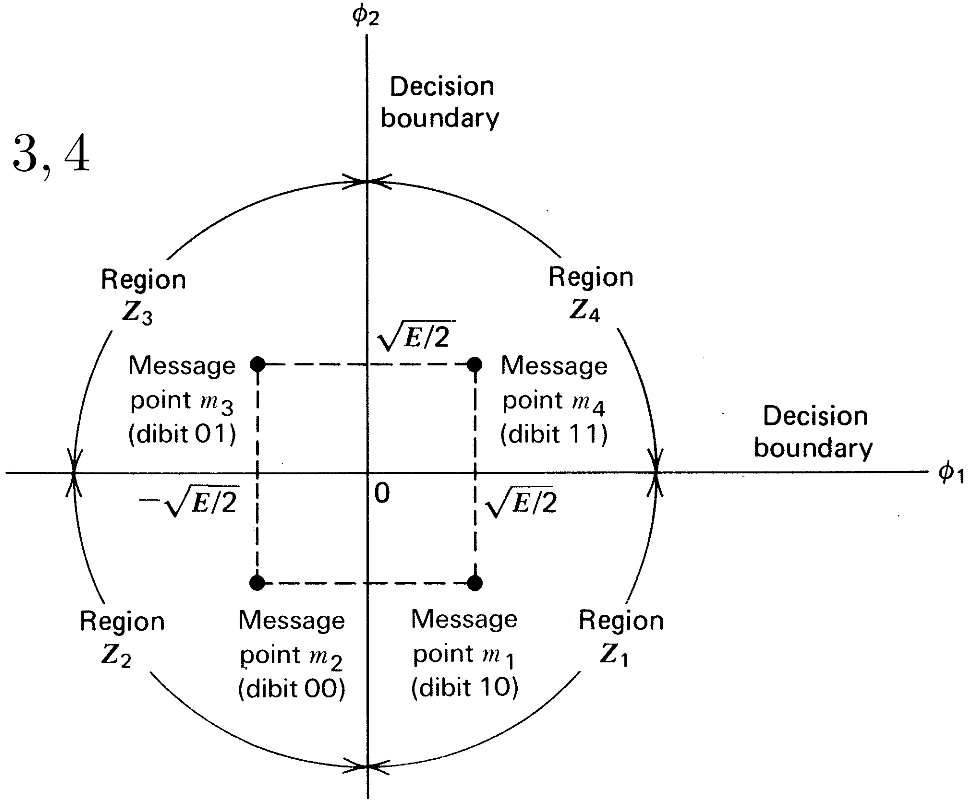


Coherent Quadri-PSK (QPSK)

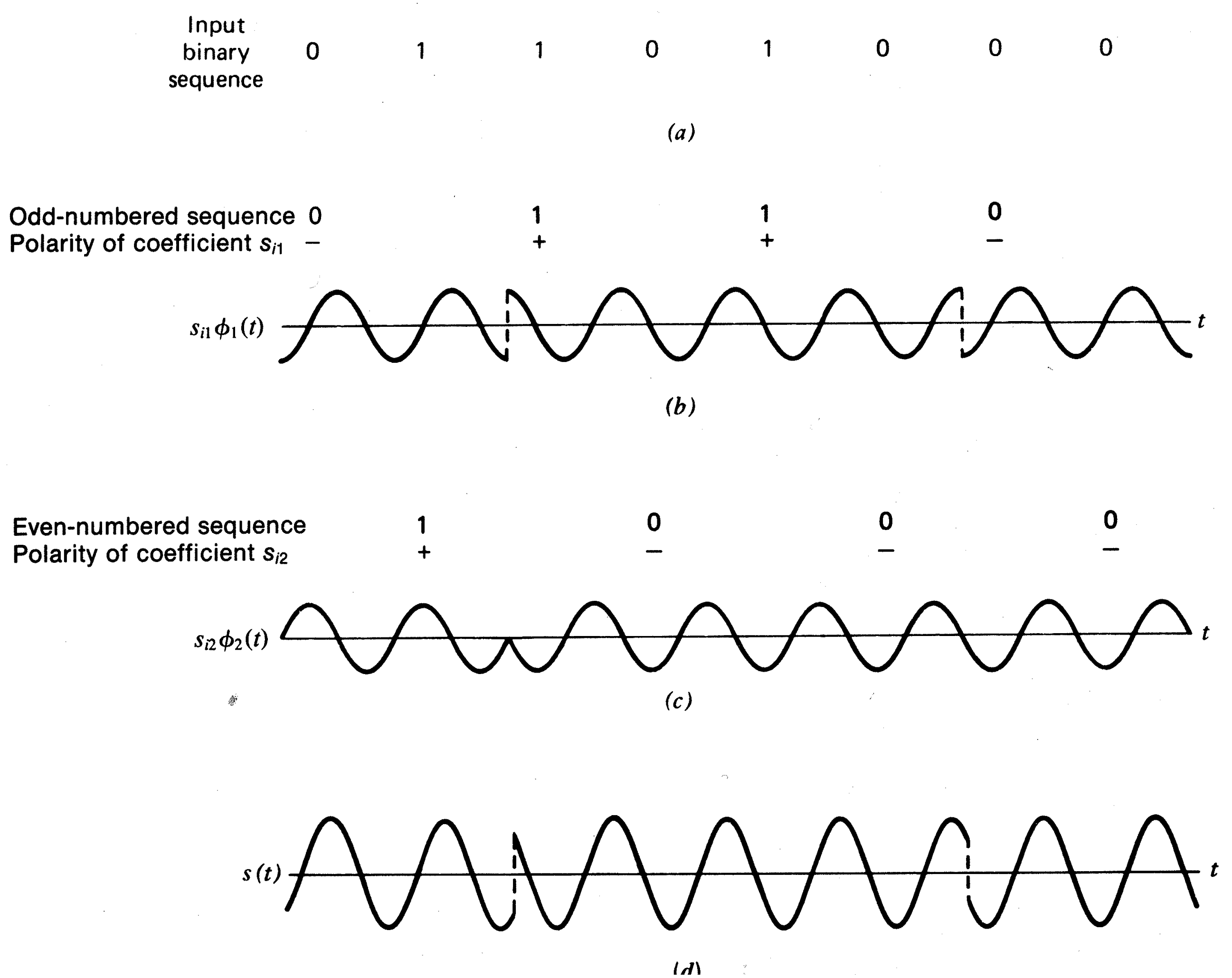
$$s_i(t) = \sqrt{\frac{2E}{T}} \cos\left[2\pi f_c t + (2i-1)\frac{\pi}{4}\right]$$

$$\mathbf{s}_i = \left[\begin{array}{c} \sqrt{E}\cos\left((2i-1)\pi/4\right) \\ -\sqrt{E}\sin\left((2i-1)\pi/4\right) \end{array} \right], i = 1, 2, 3, 4$$

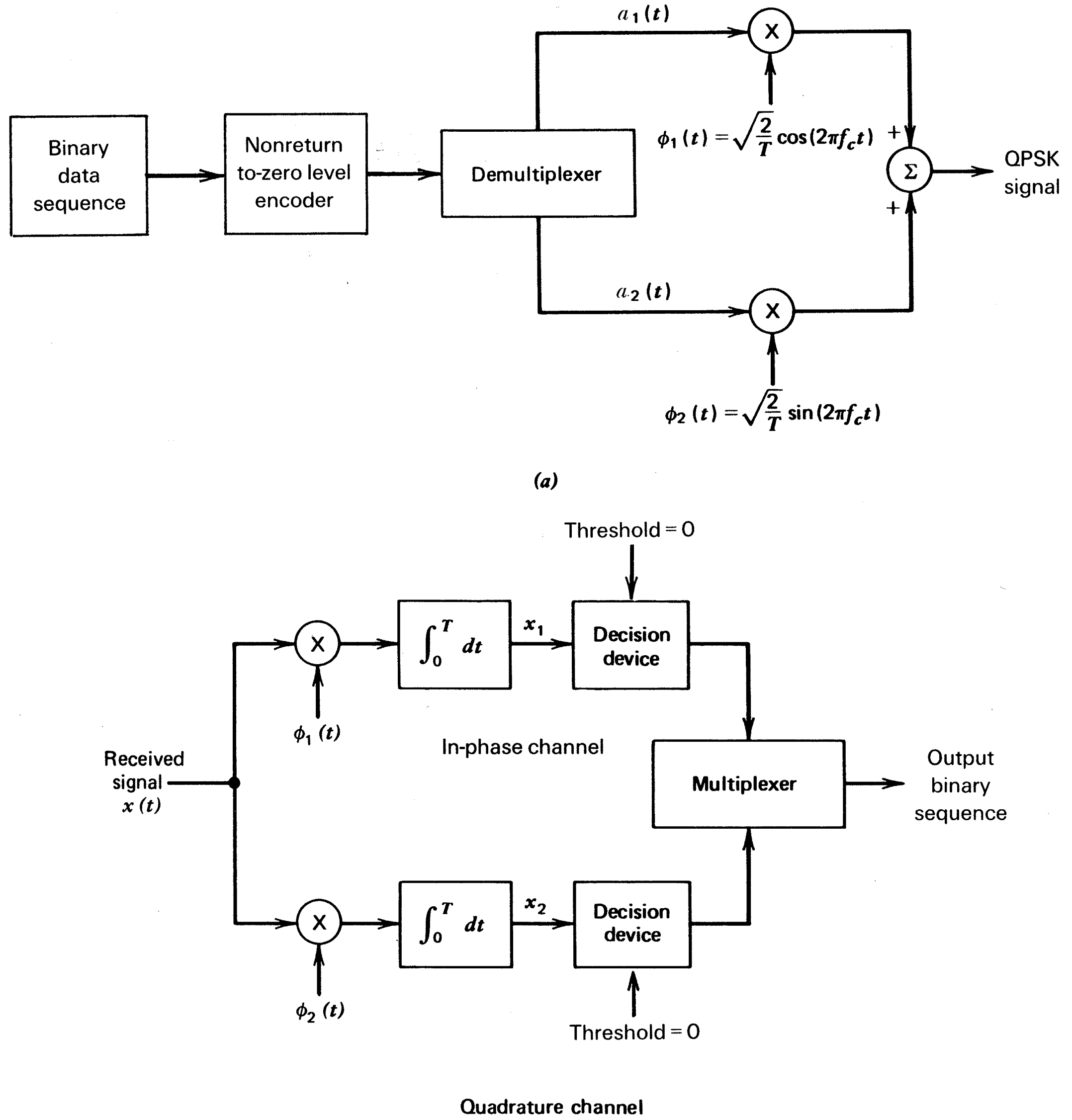
2 bits (dibit) per sent symbol



Coherent Quadri-PSK (QPSK)



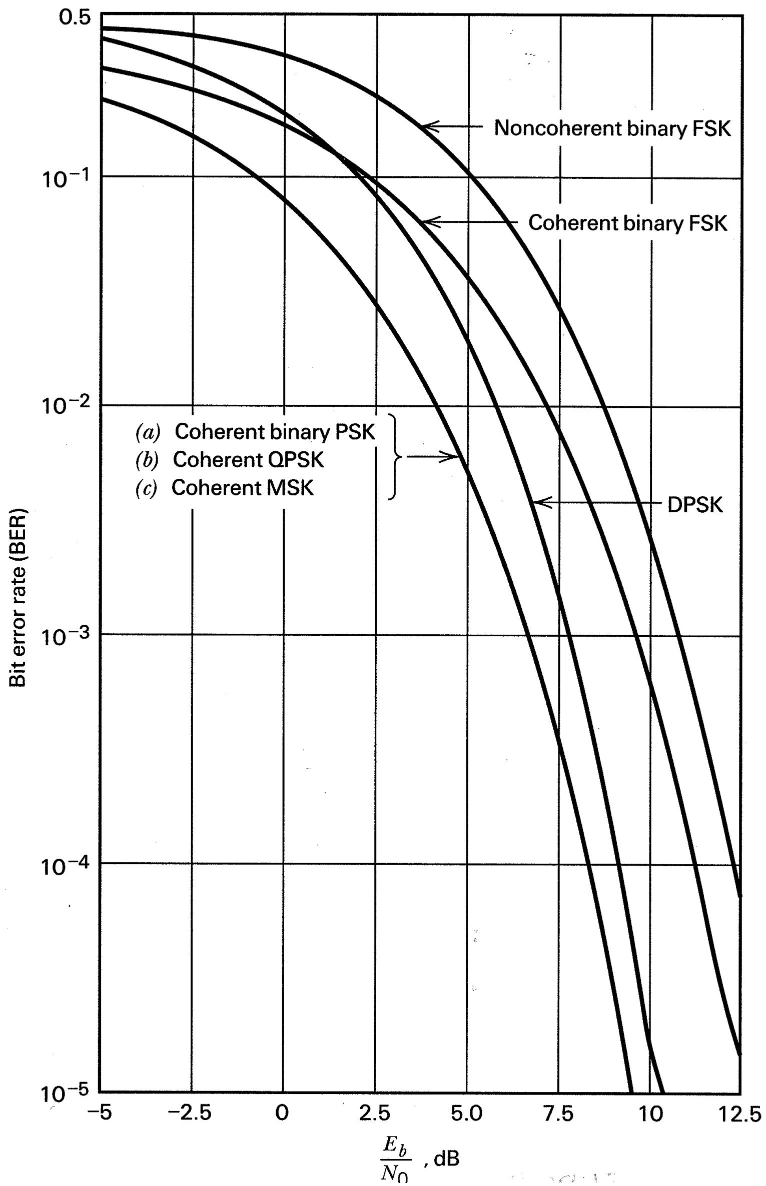
Coherent Quadri-PSK (QPSK)



Digital Modulation Schemes

Error Rate of Different Digital Modulation Schemes

Signaling Scheme	Bit Error Rate
(a) Coherent binary PSK, Coherent QPSK, Coherent MSK	$\frac{1}{2}\mathrm{erfc}(\sqrt{E_b/N_0})$
(b) Coherent binary FSK	$\frac{1}{2}\mathrm{erfc}(\sqrt{E_b/2N_0})$
(c) DPSK	$\frac{1}{2}\exp(-E_b/N_0)$
(d) Noncoherent binary FSK	$\frac{1}{2}\exp(-E_b/2N_0)$

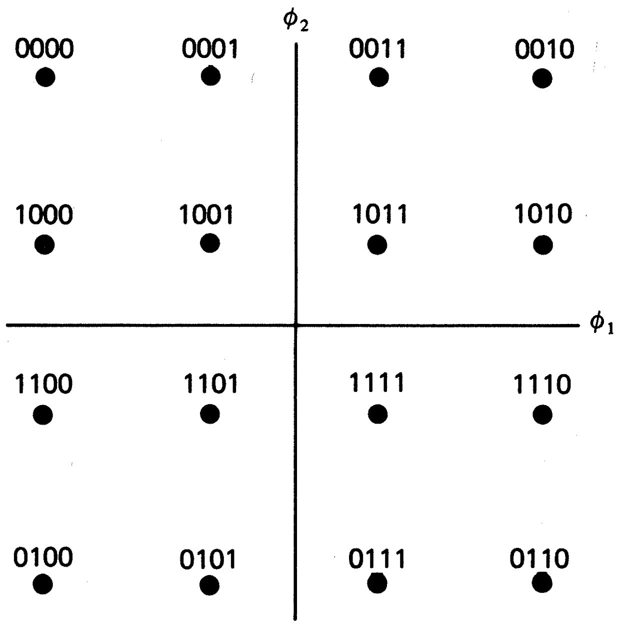


M-ary Modulation

M-ary signalling schemes are preferred over binary schemes when the requirement is to conserve bandwidth at the expense of increased power.

In **M-ary quadrature amplitude modulation (M-ary QAM)** the in-phase and quadrature components are allowed to be independent.

$$s_i(t) = \sqrt{\frac{2E_0}{T}} a_i \cos(2\pi f_c t) + \sqrt{\frac{2E_0}{T}} b_i \sin(2\pi f_c t)$$



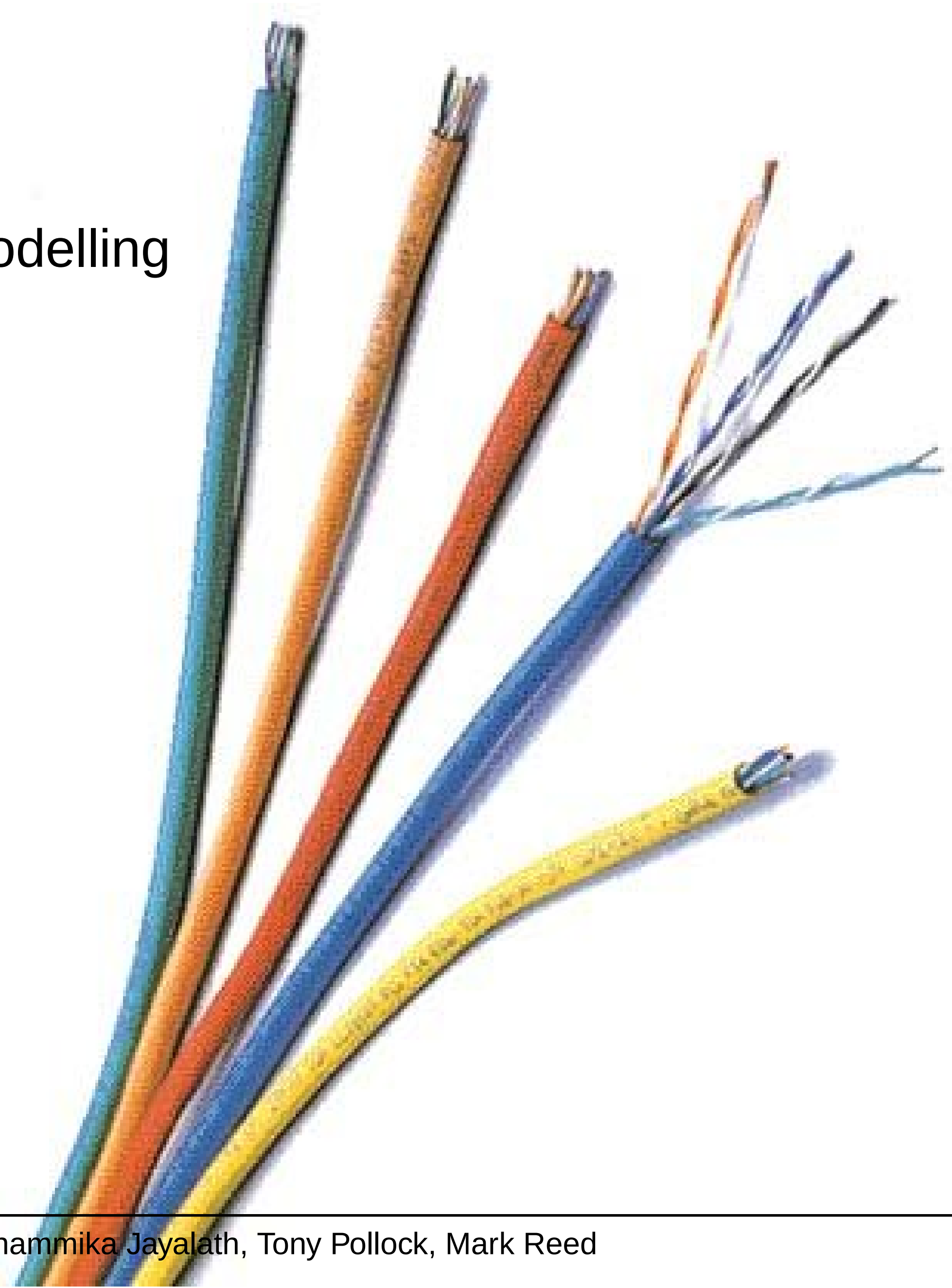
Outline

Wireless Channels :

- Wired vs Wireless
- Channel Characteristics and Modelling
- Channel Capacity

Communications Systems :

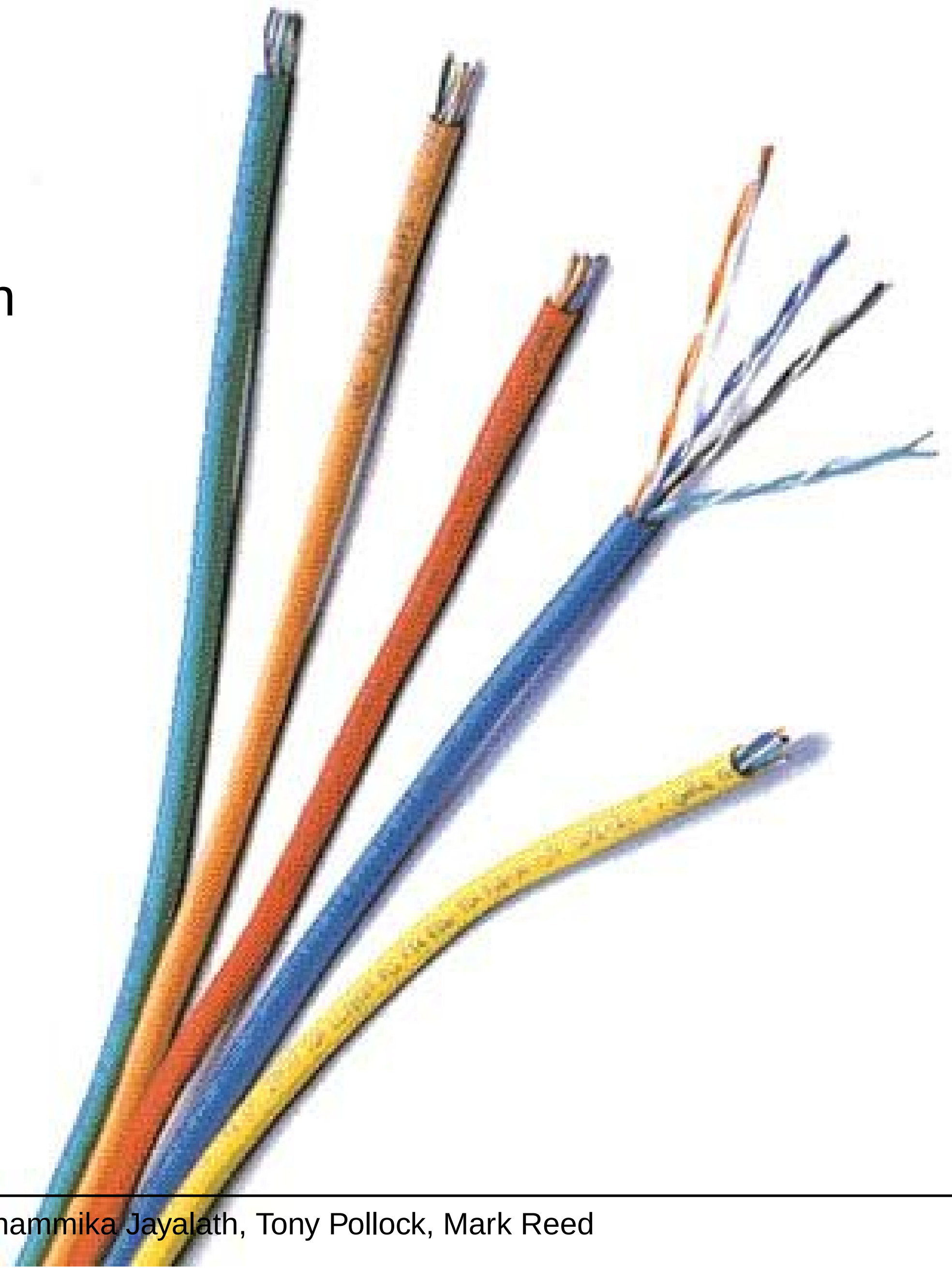
- Channel Coding and Decoding
- Modulation and Demodulation
- **Equalization**



Communications Systems

Equalization :

- Zero Forcing Equalization
- Adaptive Equalization
- Decision-Feedback Equalization
- Eye Patterns



Equalization

Equalization compensates for intersymbol interference (ISI) created by multipath within time dispersive channels.

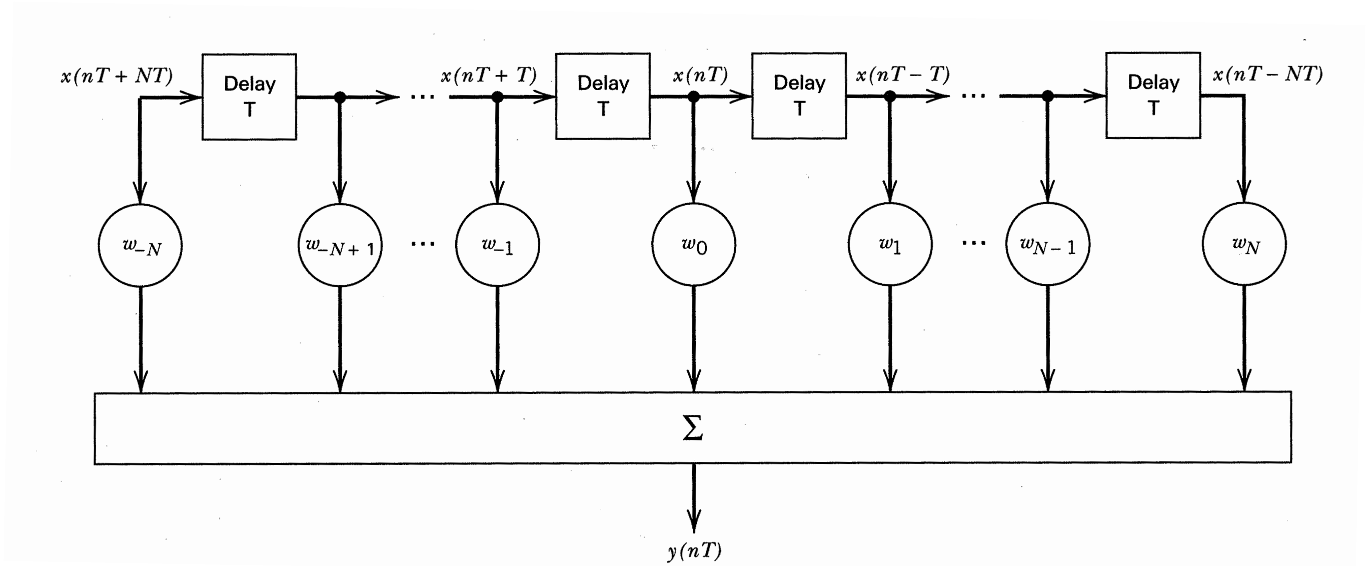
If the modulation bandwidth exceeds the coherence bandwidth of the channel, ISI occurs and modulation pulses are spread in time.

ISI is the major obstacle to high speed data transmission over wireless channels.

Equalizers in the receiver compensate for the average range of expected channel amplitude and delay characteristics.

In general equalizers need to be **adaptive** since the channel is generally unknown and time varying.

Taped-Delay-Line Filter



For symmetry the total number of taps is chosen to be $(2N+1)$: $\{w_{-N}, \cdots, w_N\}$

The impulse response of the filter is therefore

$$g(t) = \sum_{k=-N}^{N} w_k \delta(t - kT)$$

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Zero-Forcing Equalizer

Consider a ISI channel with impulse response $h(t)$. Let $p(t)$ denote the impulse response of the equalized channel

$$\begin{aligned} p(t) &= h(t) \star g(t) \\ &= \sum_{k=-N}^{N} w_k h(t - kT) \end{aligned}$$

Sampling at $t = nT$ gives the discrete convolutional sum

$$p(nT) = \sum_{k=-N}^{N} w_k h((n-k)T)$$

To eliminate intersymbol interference completely we need to define the overall pulse $p(t)$.

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Zero-Forcing Equalizer

For distortionless transmission we require the pulse at $n = 0$ to be free from ISI due to the overlapping tails of all the other pulse contributions at $n \neq 0$ **(Nyquist criterion)**:

$$p(nT) = \begin{cases} 1, & n = 0 \\ 0, & n \neq 0 \end{cases}$$

we only have $(2N+1)$ adjustable coefficients, thus we can only approximate

$$p(nT) = \begin{cases} 1, & n = 0 \\ 0, & n = \pm 1, \pm 2, \ldots, \pm N \end{cases}$$

Zero-Forcing Equalizer

The taped-delay-line equalizer taps w_k must satisfy the set of $(2N+1)$ simultaneous equations

$$\sum_{k=-N}^{N} w_k h((n-k)T) = \begin{cases} 1, & n = 0 \\ 0, & n = \pm 1, \pm 2, \ldots, \pm N \end{cases}$$

The zero-forcing equalizer is optimum in the sense that it minimises the peak distortion (ISI).

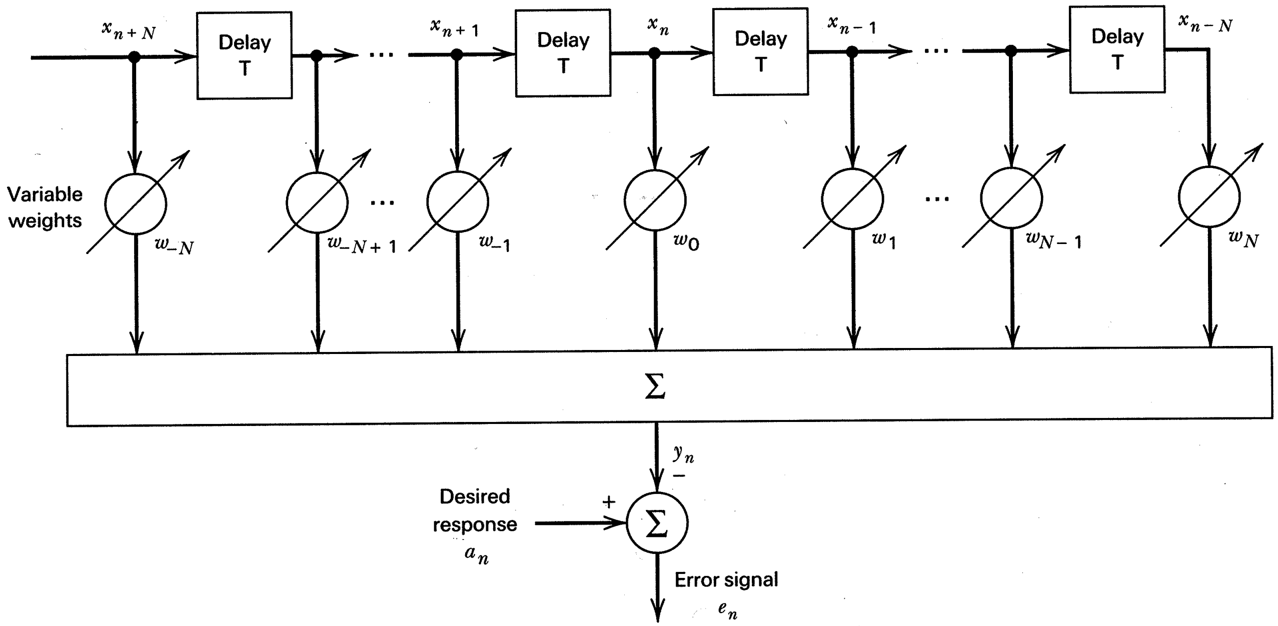
The longer the equalizer the more closely the equalized system approaches the distortionless channel case.

Although this equalizer is relatively simple to implement, it requires the channel coefficients, however in a communications environment the channel is time varying and the equalizer needs to be **adaptive**.

Adaptive Equalization

An **adaptive equalizer** is one which adjusts itself continuously and automatically by operating on the input signal.

$$y_n = \sum_{k=-N}^{N} w_k(n) x_{n-k}$$



Adaptive Equalization - LMS

Adaption is achieved by observing the error e_n between the desired pulse shape and the actual pulse shape at the sample instants, then estimate the direction in which the tap weights should change.

Least-mean-square (LMS) algorithm moves the taps in the steepest descent direction of the error surface $\mathcal{E} = E\{e_n^2\}$, i.e.

$$w_k(n+1) = w_k(n) - \mu \frac{1}{2} \frac{\partial \mathcal{E}}{\partial w_k}$$

where μ is the *step-size parameter* and

$$\frac{\partial \mathcal{E}}{\partial w_k} = -2R_{ex}(k) \approx -2e_n x_{n-k}$$

Adaptive Equalization - LMS

There are two modes of operation for the adaptive equalizer:

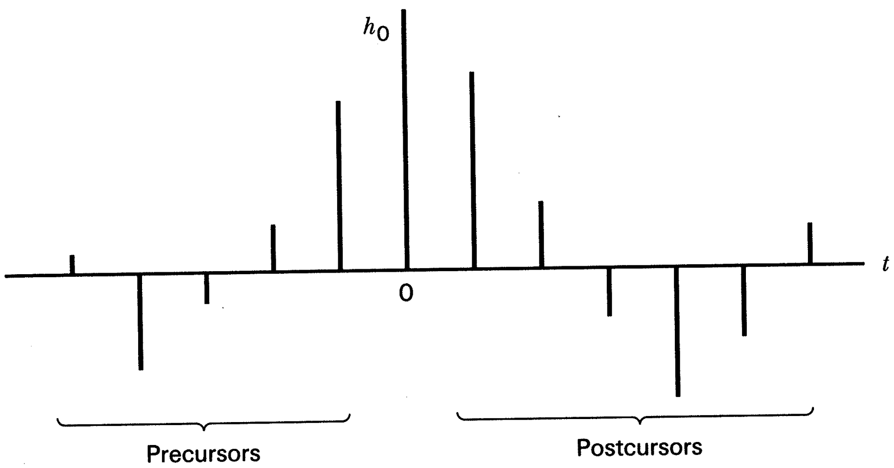
Training mode: a known sequence is transmitted and a synchronized version of this signal is generated at the receiver and is applied to the adaptive equalizer as the desired response - the taps are then adjusted according to the LMS algorithm.

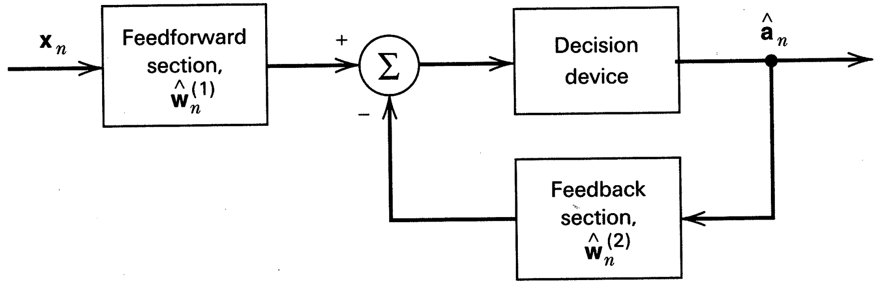
Decision directed mode: the equalizer operates in normal LMS mode and is able to track relativity slow variations in channel characteristics. The ability to track depends on the step size, however large step size often incurs an excess mean-square error.

Adaptive Equalization - DFE

Decision-Feedback Equalization (DFE) uses data decisions made on the basis of *precursors* to take care of the *postcursors*.

$$y_n = h_0 x_n + \underbrace{\sum_{k<0} h_k x_{n-k}}_{\text{precursors}} + \underbrace{\sum_{k>0} h_k x_{n-k}}_{\text{postcursors}}$$





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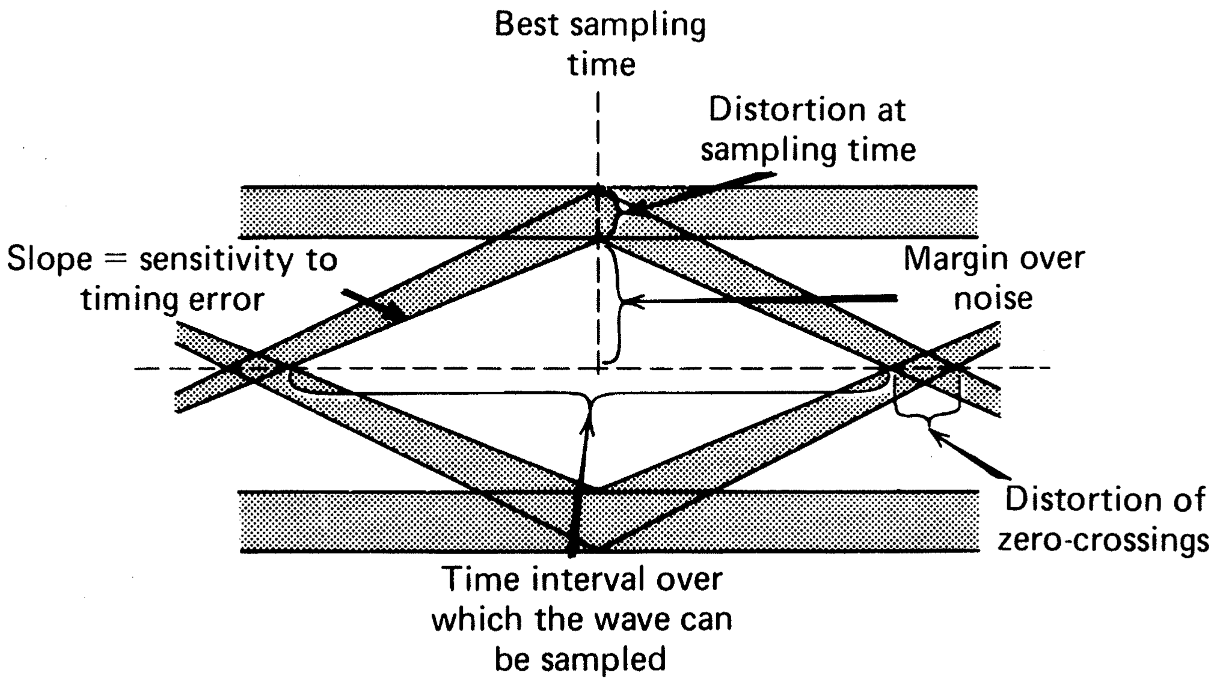
Adaptive Equalizers

Algorithm	Number of Multiply Operations	Advantages	Disadvantages
LMS Gradient DFE	$2N + 1$	Low computational complexity, simple program	Slow convergence, poor tracking
Kalman RLS	$2.5N^2 + 4.5N$	Fast convergence, good tracking ability	High computational complexity
FTF	$7N + 14$	Fast convergence, good tracking, low computational complexity	Complex programming, unstable (but can use rescue method)
Gradient Lattice	$13N - 8$	Stable, low computational complexity, flexible structure	Performance not as good as other RLS, complex programming
Gradient Lattice DFE	$13N_1 + 33N_2 - 36$	Low computational complexity	Complex programming
Fast Kalman DFE	$20N + 5$	Can be used for DFE, fast convergence and good tracking	Complex programming, computation not low, unstable
Square Root RLS DFE	$1.5N^2 + 6.5N$	Better numerical properties	High computational complexity

Eye Diagram

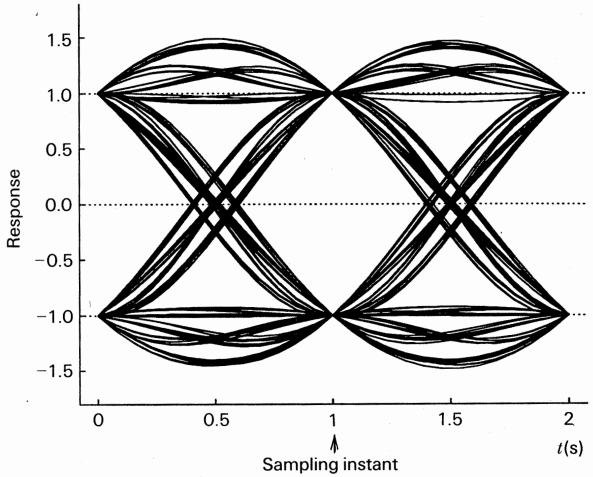
Synchronized superposition of all possible realizations of the signal of interest.

Useful experimental tool for evaluation of the impairments on the overall system performance in an operational environment.

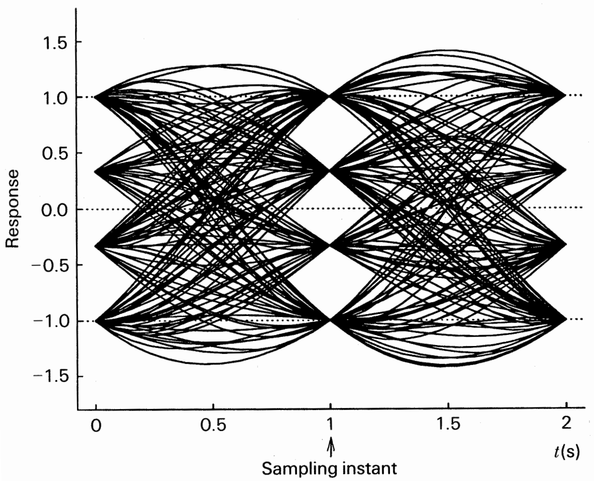


Eye Diagram

no bandwidth limitation

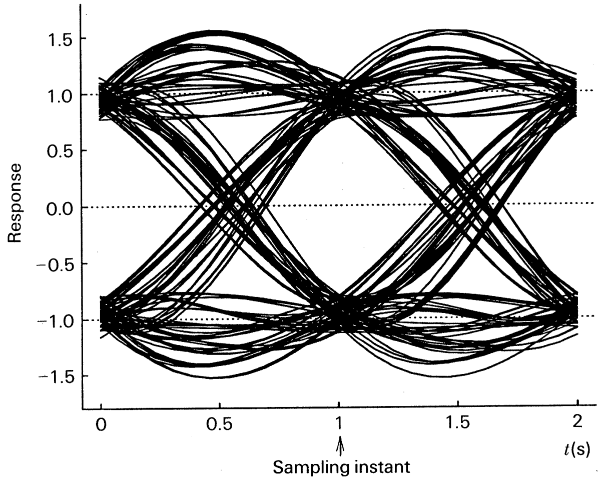


(a) $M = 2$

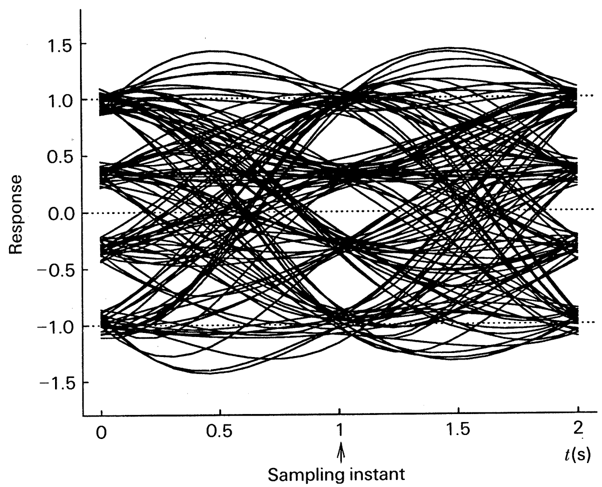


(b) $M = 4$

bandwidth-limited channel

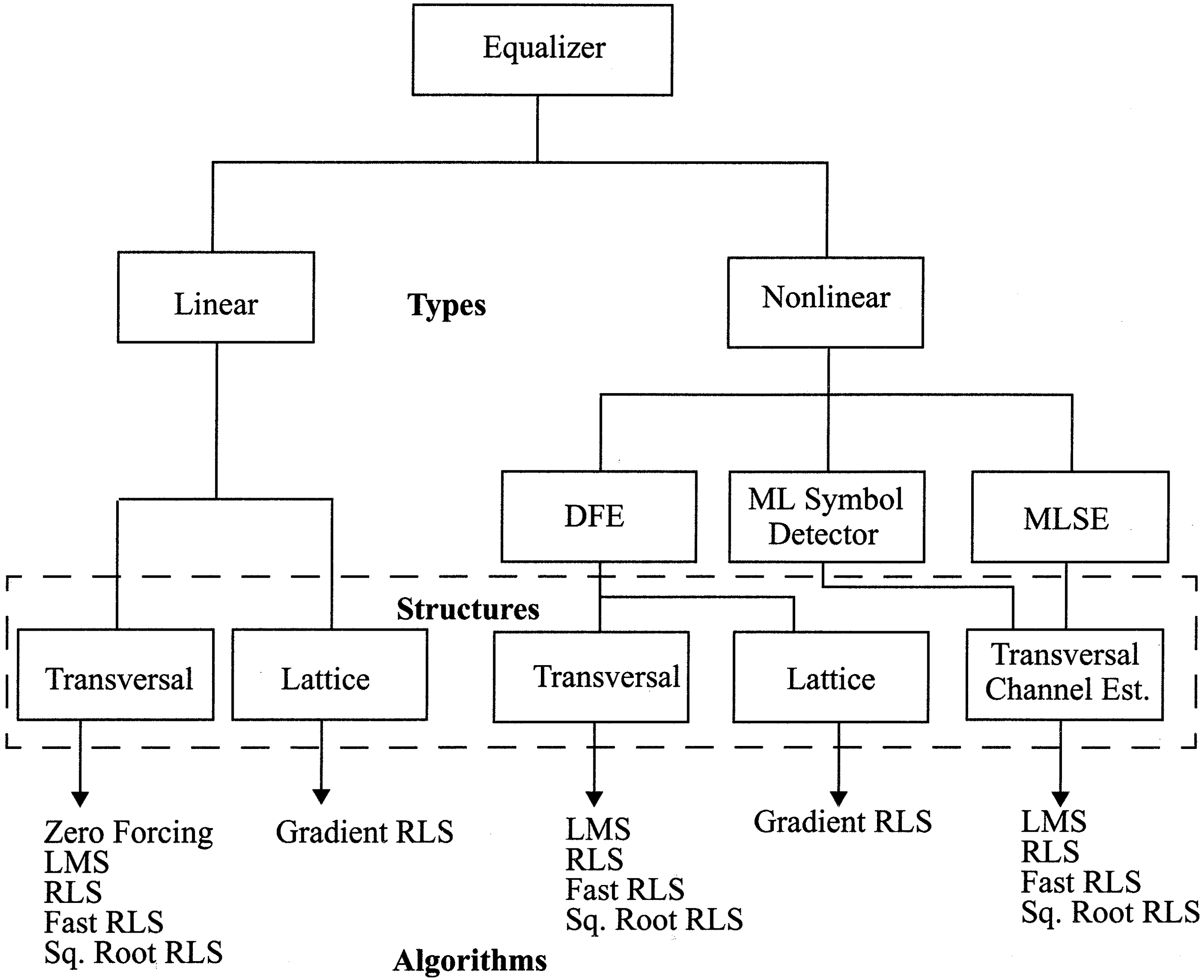


(a) $M = 2$



(b) $M = 4$

Equalizers Overview



Overview of Wireless Communications
Multi-user Channels and Multi-access Channels

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March 2004

Overview

- Multiuser Problem
- Multi-access Methods (TDMA, CDMA, FDMA, SDMA)
- Current Multi-access Standards
- Wireless Channel Conditions
- Channel Modelling (Linear Algebra)
- Cellular Network Issues
- Example CDMA S-UMTS Hardware System
- Detection Criteria
- Multiuser DS/CDMA System (Turbo CDMA)
- Interference Canceller Design
- DS/CDMA Acquisition Problem
- MAP Algorithm

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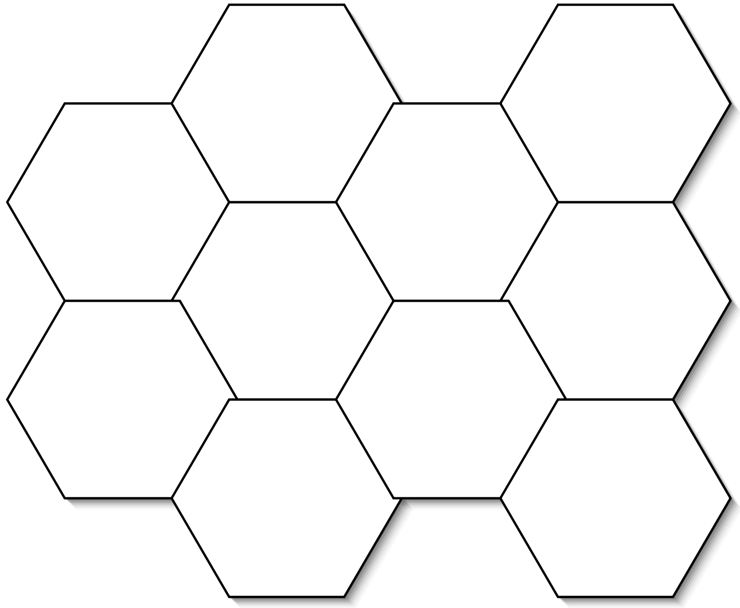
Multiple user problem

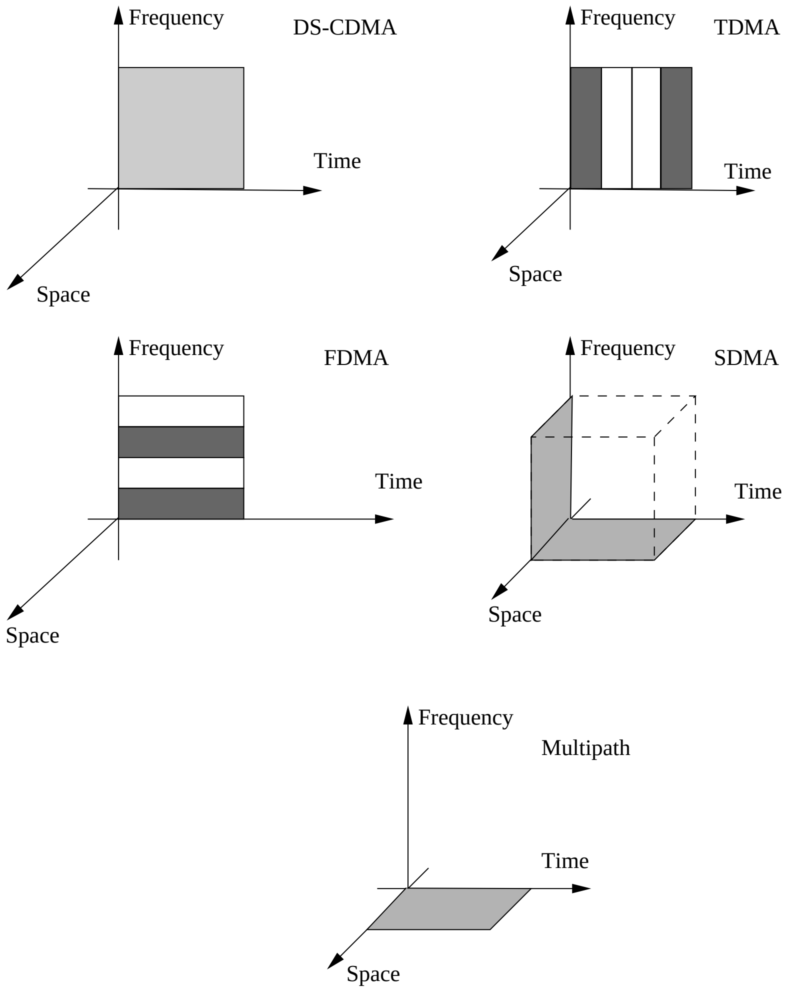
- Scenario - Mobile Communication System
- Millions of users (phones/terminals)
- Thousands of basestations (one for each cell)
- How do you separate users?

Ans: Cellular concept and Time, Space, and Frequency



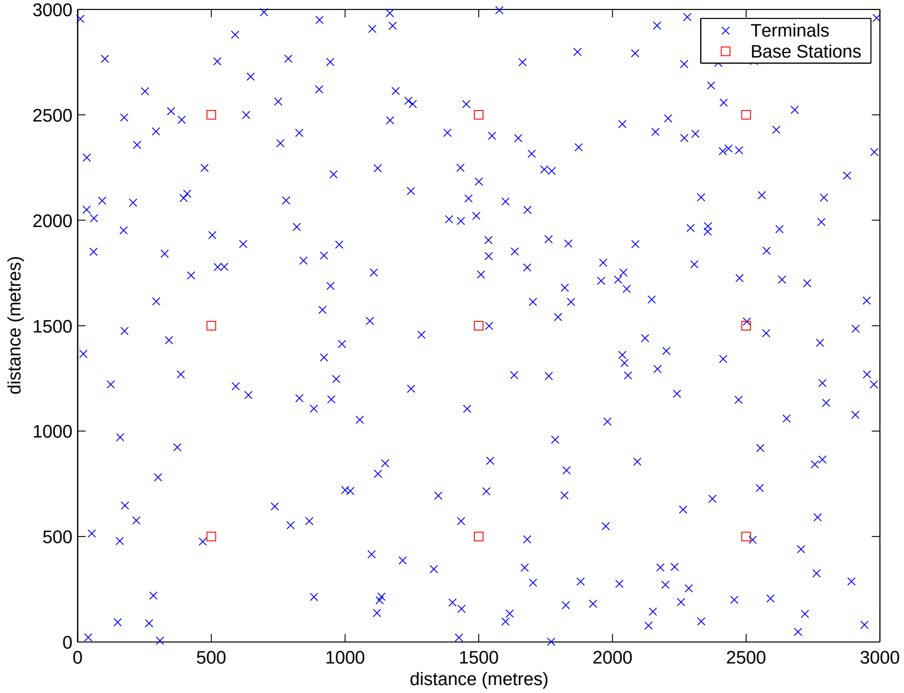
Diversity - Space, Time, and Frequency

- Three dimensions of seperation
- Direct-Sequence Code-Division Multiple-Access (DS-CDMA)
- Time-Division Multiple-Access (TDMA)
- Frequency-Division Multiple-Access (FDMA)
- Space-Division Multiple-Access (SDMA)
- Multipath (exploiting diversity)



Motivating Example- Interference Cancellation gains

- In DS/CDMA uplink systems interference limits the user loading
- Conventional Receivers can support users(K)/processing gain(N) = 0.3
- Turbo CDMA receivers (Iterative MUD) can support K/N=2 loading



- A Multiuser Detector means:-
  - More users can be supported - more revenue for the service provider
  - Each mobile user has a longer range- fewer basestations needed
  - Each mobile user has a longer battery life - better conditions for users
  - Each mobile user has better reception quality- better perception by users

Overview

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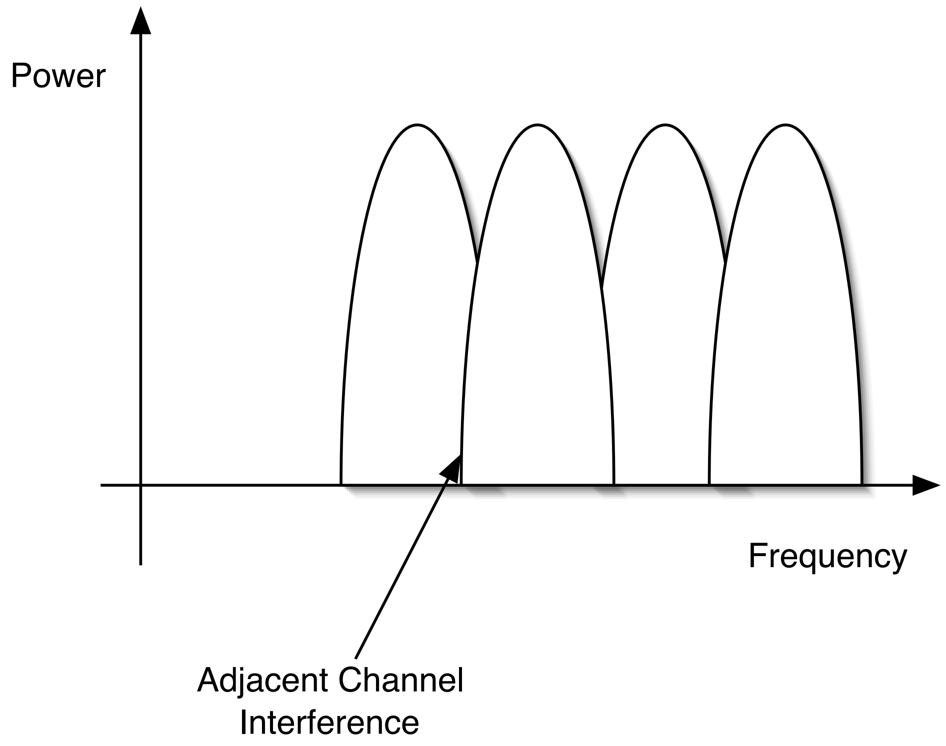
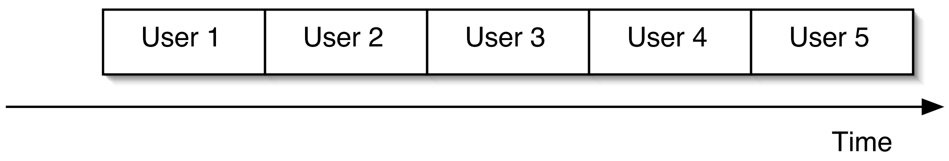
FDMA

- Users are separated by frequency
- System suffers from Adjacent Channel Interference (ACI)
- Frequency offsets between users cause problems
- Transmitter filter design is important
- Used in GSM System (TDMA and FDMA)
- Commonly used in Satellite Systems



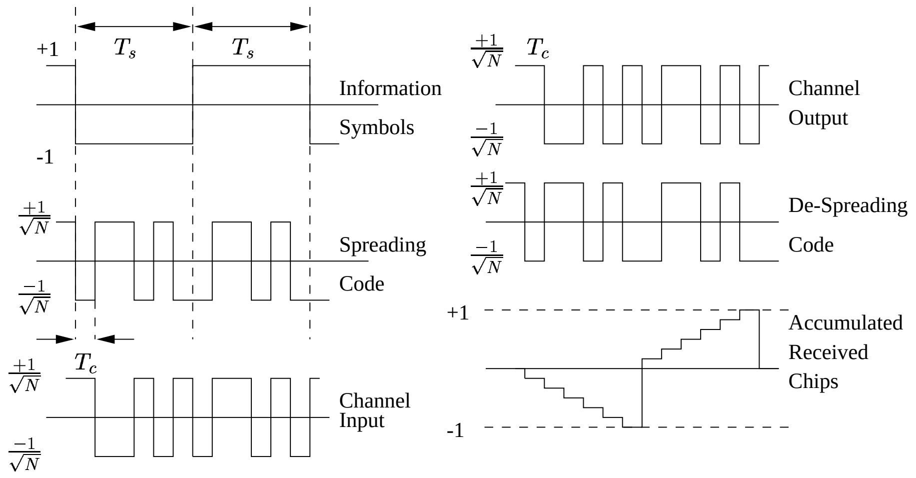
TDMA

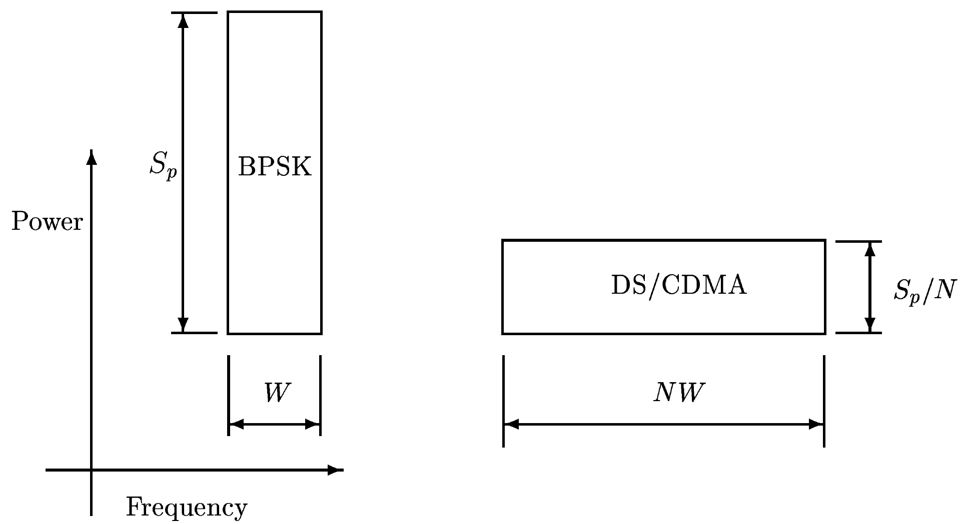
- Users are separated in time
- All users share the same frequency
- Strict timing control is needed
- Used in GSM mobile phone system



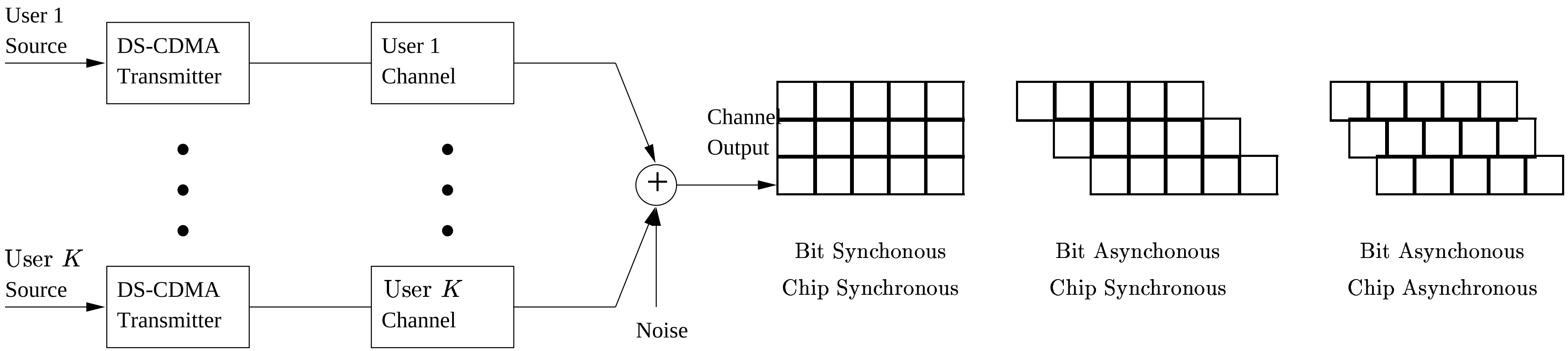
CDMA

- For each data symbol a code of multiple chips is generated and the data is “spread”
- The receiver knows the code and can “de-spread”
- Advantages include
  - Tolerance to Carrier interference
  - Diversity over time and frequency
  - Ability to allow multiple users (multiple users stack up)





CDMA - con't



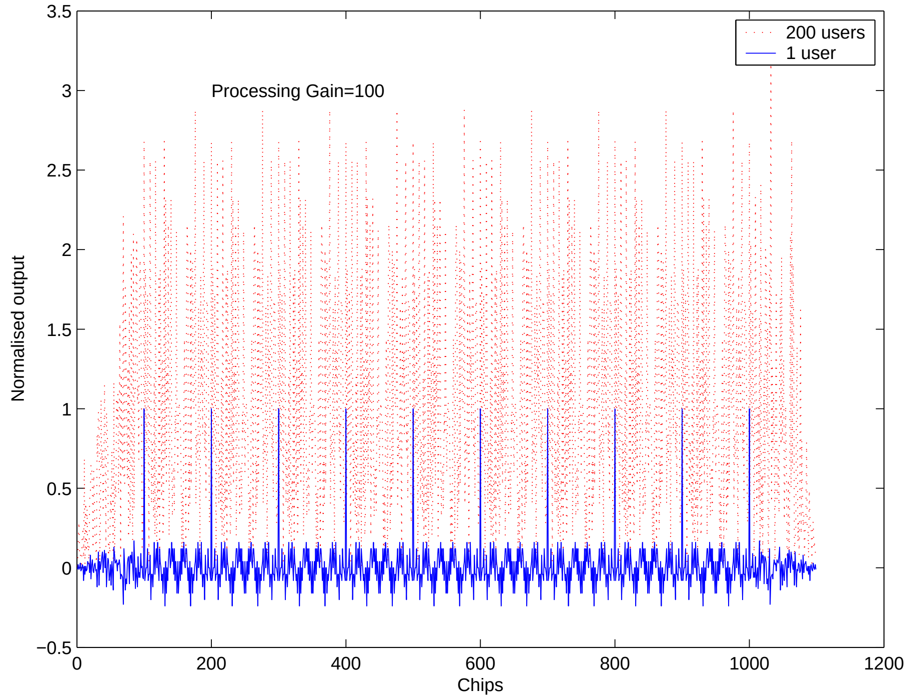
- For each user there is a separate channel (in uplink MS to BS)
- All users signals are added (combining at antenna)
- Noise is added also (Low Noise Amplifier noise in Receiver)
- Combined signal is passed to receiver
- Modelling of multi-user asynchronism can have different forms

An Analogy to CDMA

- Attending an outdoor orchestra recital
- Sounds are coming from multiple sources (Co-channel Interference)
- Each Instrument has its own sounds (form of code)
- If one instrument is too loud then the others are drowned out (Near-far problem)
- Noise from other sources can also drown out what you hear
- Bad position, two time delayed signals making it difficult to hear (multipath)
- Cup your hands to hear the orchestra more clearly (Antenna gain)

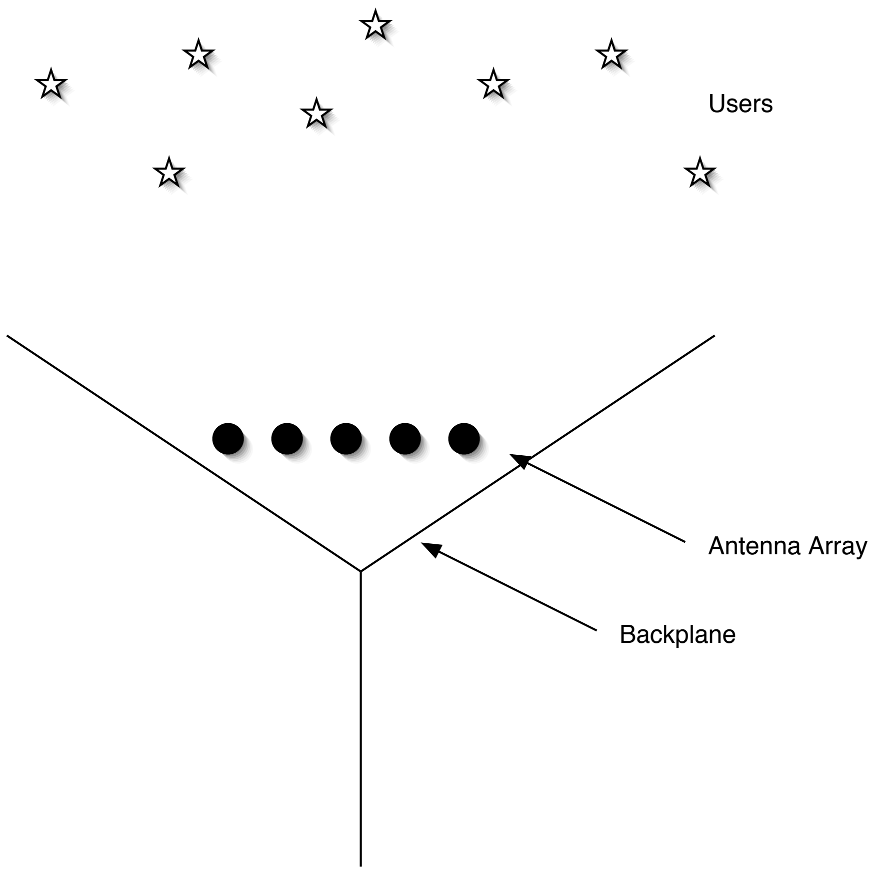
Multiple Access Interference (MAI) in DS/CDMA

- Figure shows Correlation peaks for single user and multiple users
- Single user is easily identifiable
- Multiuser is not possible to see timing location, due to MAI
- Signal processing mechanisms are needed to reduce MAI



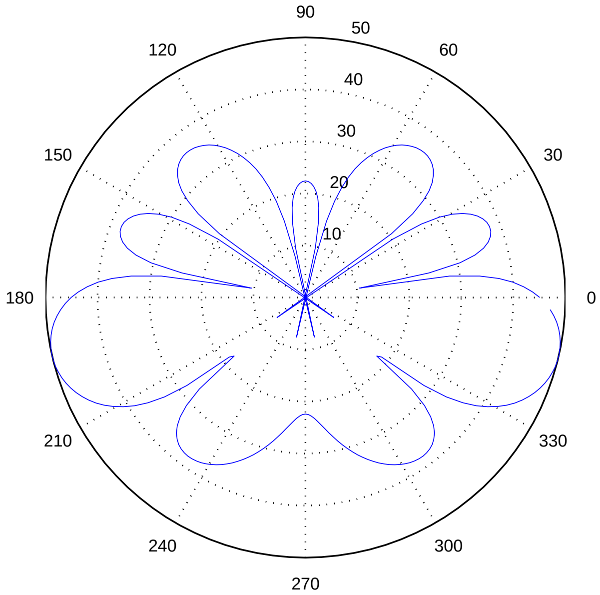
SDMA (1)

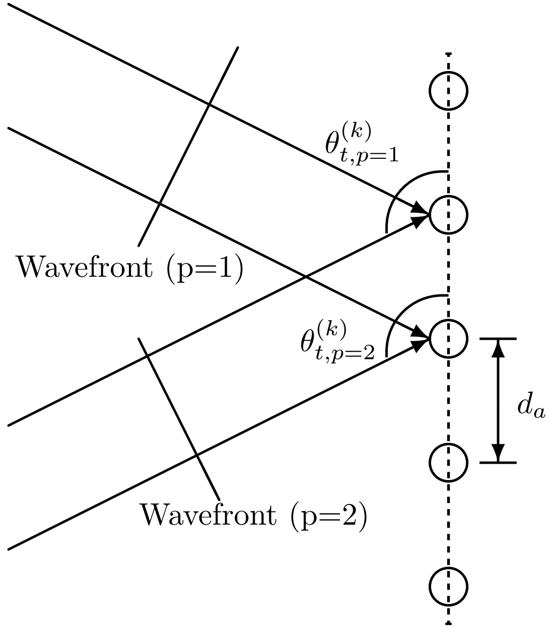
- Multi user system with Narrowband BPSK (only spatial separation)
- Direction of Arrival (DOA) determines separation of users
- $L = 5$ antenna elements (Uniform Linear Array - ULA), 120 degrees sector
- Spatial Model - Direction of arrival (DOA) is uniformly distributed over sector
- GSM Basestations could check DOA and allocate frequency/time slot position



3 Sector SDMA Representation

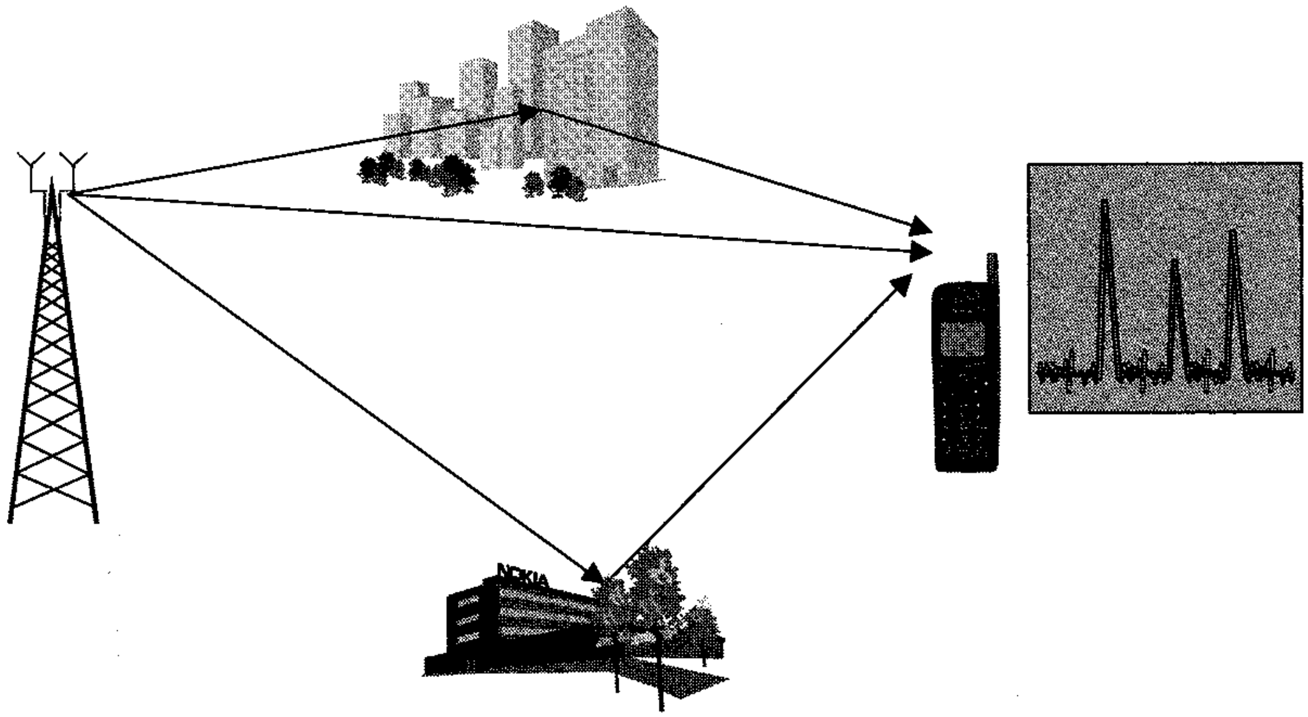
SDMA (2)





- Beampattern for a 5 element ULA Antenna
- 45 degrees pointing direction
- Steering Vector $\mathbf{e} = \begin{bmatrix} 1 & \exp\left\{-j\pi l \sin(\theta^{(k)})\right\} & \cdots & \exp\left\{-j\pi (L-1) \sin(\theta^{(k)})\right\} \end{bmatrix}^T$

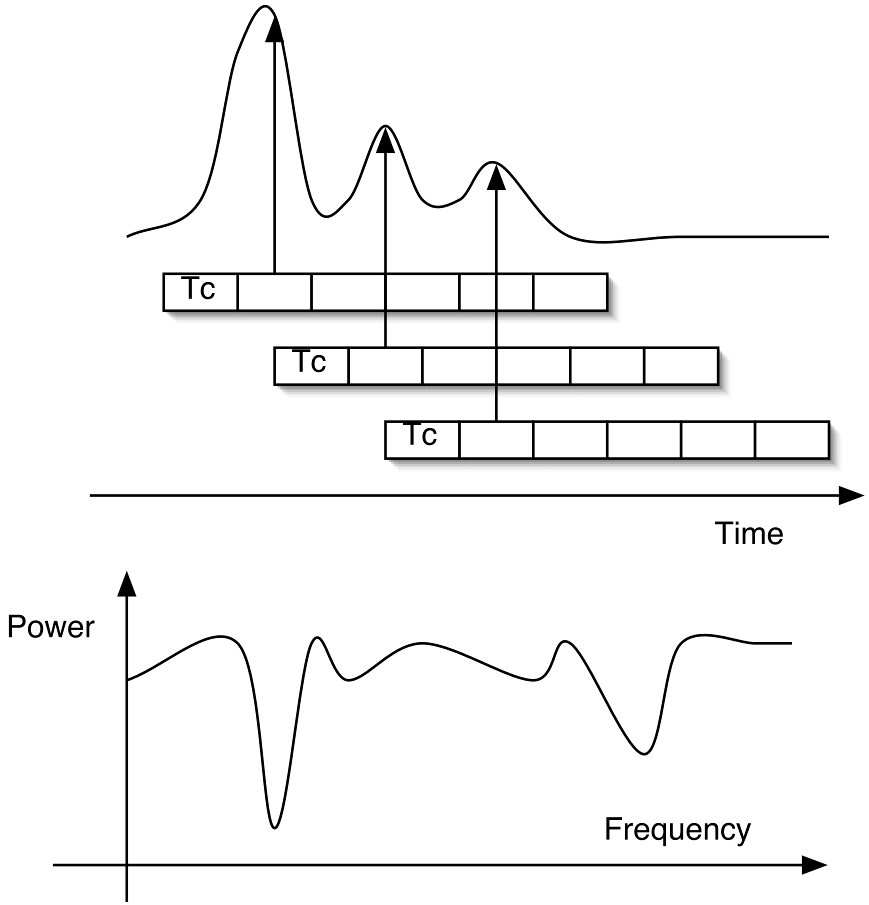
Multipath



- Multipath is created from reflections from buildings etc.
- Multipath can be both destructive and constructive to signal
- DS/CDMA systems can take advantage of multipath diversity

Multipath

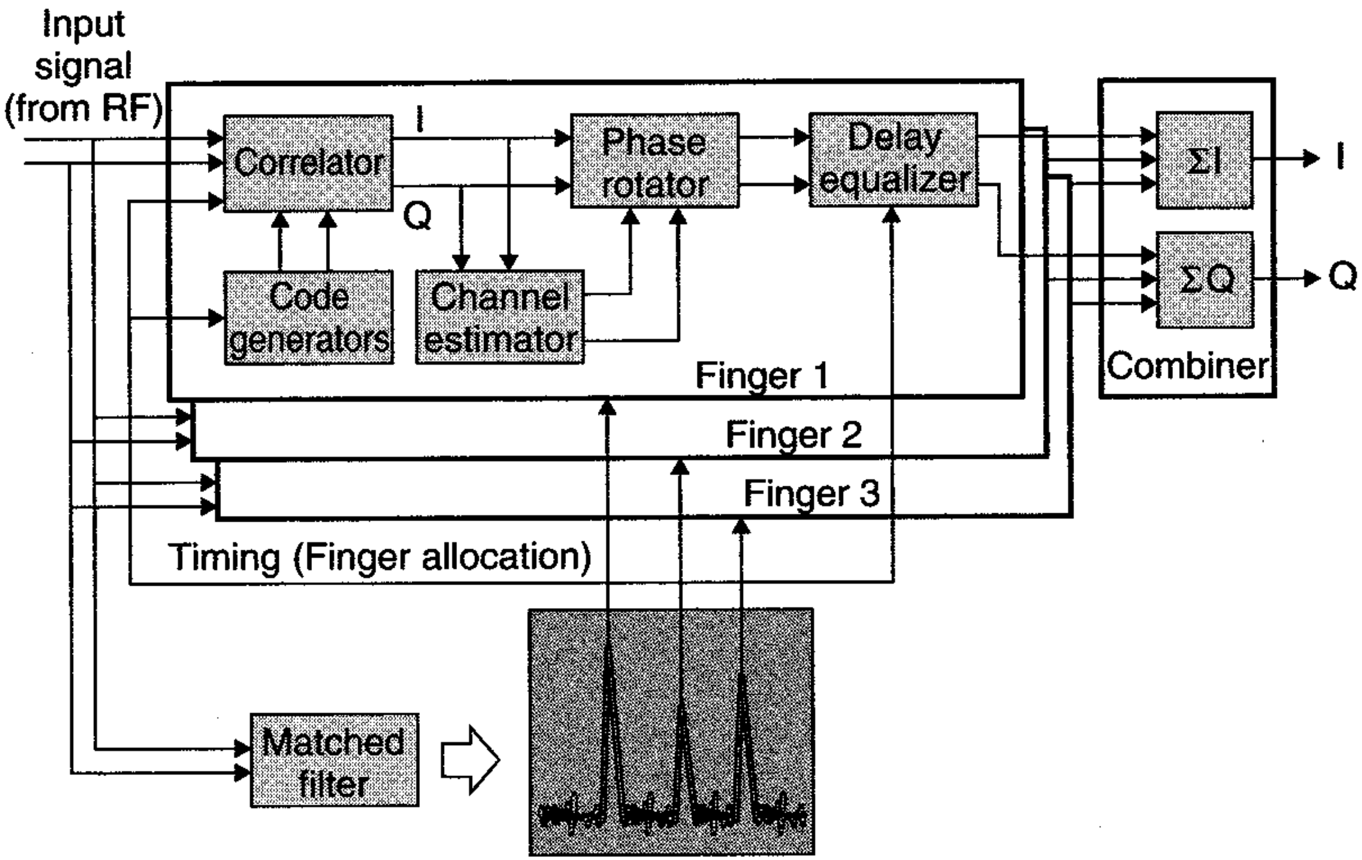
- In Systems where the multipath time separation is greater than the symbol/chip time (i.e. DS/CDMA), then diversity gains can be achieved by combining multiple multipaths
- UMTS - 3.84MHz chips (260ns chips), therefore multipaths greater than 260 ns can be combined
- Generates Frequency Selective Channel
- Typical Mulipath representation



- As the multipath separation is greater than one chip (Tc) the multipath can be resolved
- Optimum combining is Maximum Ratio Combining $\sum_p m_p^2$

Multipath- Compensation with a RAKE Receiver

- RAKE receiver has multiple “fingers” to combine the multipath components



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- Multi-access Methods (TDMA, CDMA, FDMA, SDMA)
- **Current Multi-access Standards**
- Wireless Channel Conditions
- Channel Modelling (Linear Algebra)
- Cellular Network Issues
- Example CDMA S-UMTS Hardware System
- Detection Criteria
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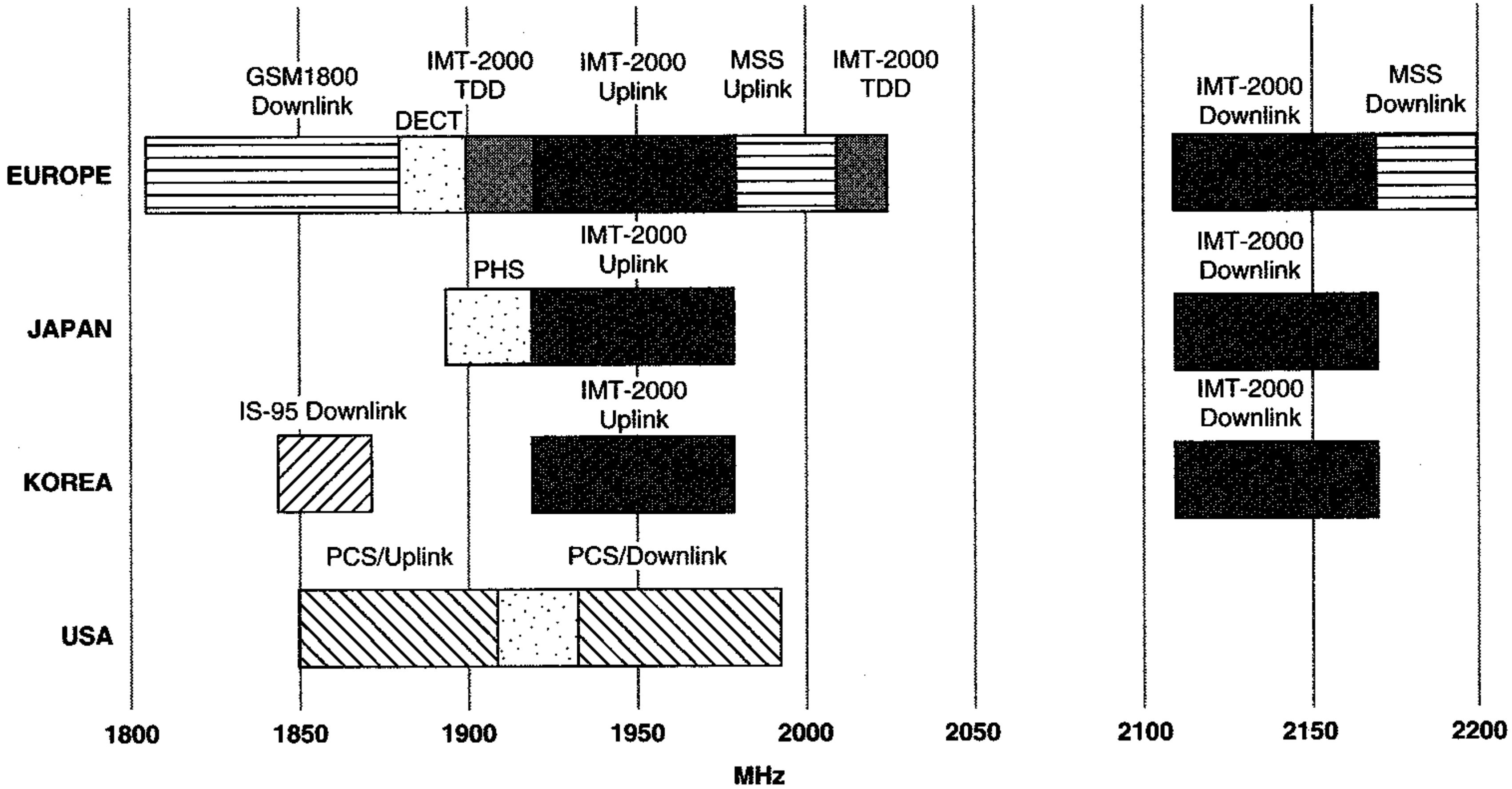
- MAP Algorithm

Overview of current Multi-access Standards

- CDMA - high mobility users (up to 200km/hr)
  - IS-95 , 10kbps per user, 1.22 MHz bandwidth, 2G system
  - 3GPP (UMTS), up to 384kbps per user, 3.88 MHz b/w, 3G system
- TDMA/FDMA - high mobility users (up to 200km/hr)
  - GSM System, 10kbps per user, 200kHz slots, 2G system, modulation is GMSK for good spectral efficiency
- Orthogonal Frequency Digital Modulation (OFDM)
  - 802.11 Standard (WiFi/WLAN), up to 54Mbps, low mobility users
  - 802.16 Standard (Fixed Wireless Access-FWA), Mbps to the house for last mile connections, provided together with multiple antennas at Transmitter and Receiver (MIMO)
  - Further 4G systems are being developed based on OFDM and MIMO.

Spectrum allocation for 3G services



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Multiuser/Cellular Channel Conditions

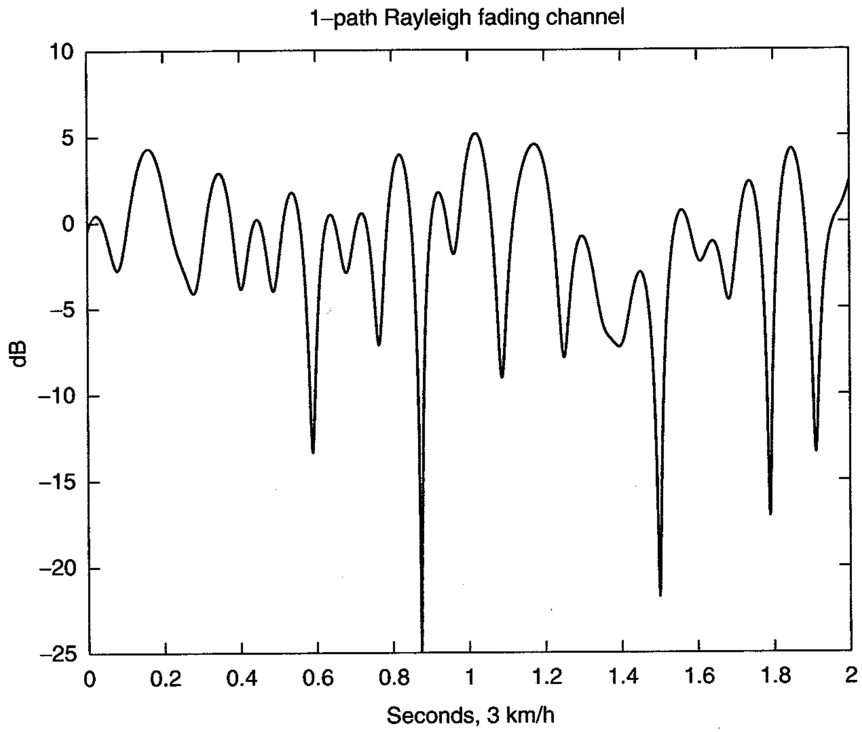
- Path Loss
- Shadowing
- Fast Fading (time selective fading)
- Multipath (frequency selective fading)
- Interference (statistics of interference)

Path Loss

- As the distance between the Transmitter and Receiver increase the path loss increases based on some rule
- The path loss coefficient depends on the environment (urban, hilly, suburban, etc.)
- For Cellular systems the greater the path loss the better the system performs as they are interference limited
- A Typical path loss channel model could be $L = r^{-\eta}$ dB
- Values in the range of $2 \leq \eta \leq 4$ are typical

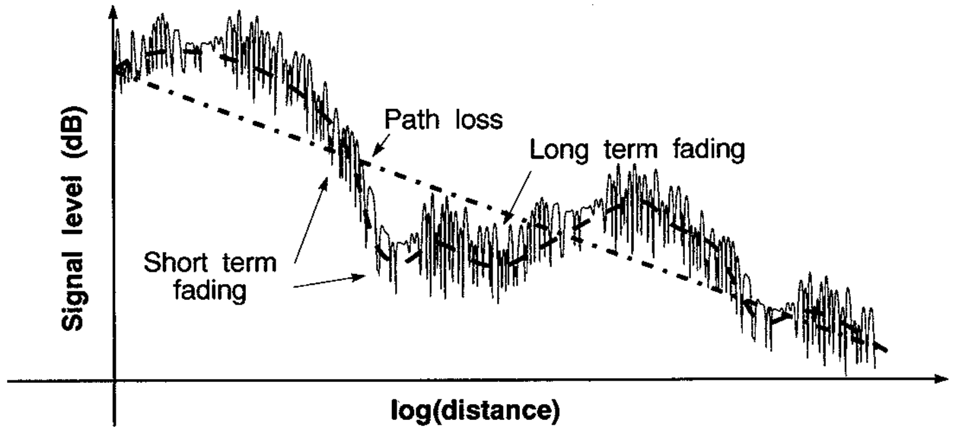
Fast Fading

- Figure shows received power over time for a given speed
- The distribution of this process is Rayleigh distributed
- The process can be modelled by the use of the “Jakes” model
- The design of mobile cellular systems takes into account the fast fading process in the standardisation process (block sizes for example)

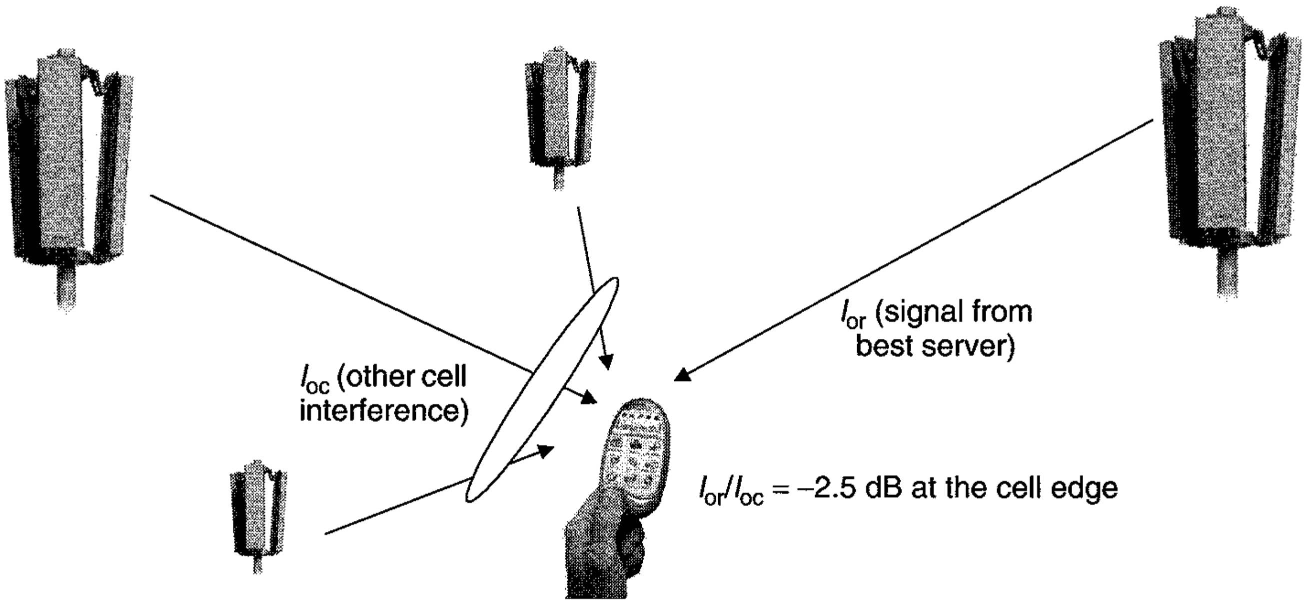


Shadowing

- A Typical Shadowing channel model could be $L = 10^{(\phi/10)}$
- ϕ has a Gaussian distribution where typical values of the standard distribution are 4dB $\leq \sigma \leq 10$dB
- Figure shows overall fading profile over distance (fast,slow and path loss)



Interference and it’s statistics(1)



- Interference comes from within the cell from other users
- Half the interference comes from neighbouring cells
- As the cell of interest knows little about the other cells this is treated as noise
- Techniques such as soft-handover reduce the effect of intra-cell interference

Interference and it’s statistics(2)

- Not all interference is the same
- Unknown interference = AWGN
- Knowledge about interference can assist interference canceller
- statistics of interference can be also useful
- Example, Transmission interference in TV Band (BushLAN Problem)
- Method of Attack, Transmissions are nearly always carriers, therefore
  - Implement carrier frequency searcher
  - track any carriers and implement nulling filter or frequency canceller circuit
- in DS/CDMA you may know code used, channel coding used, etc, etc....

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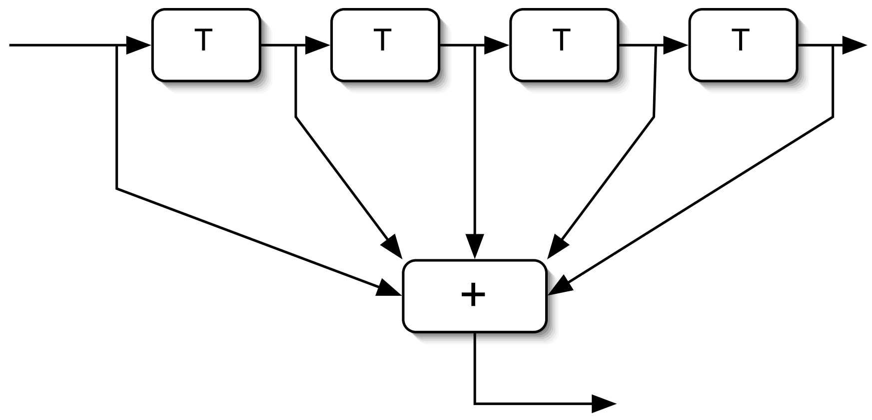
Generalised Code Construction

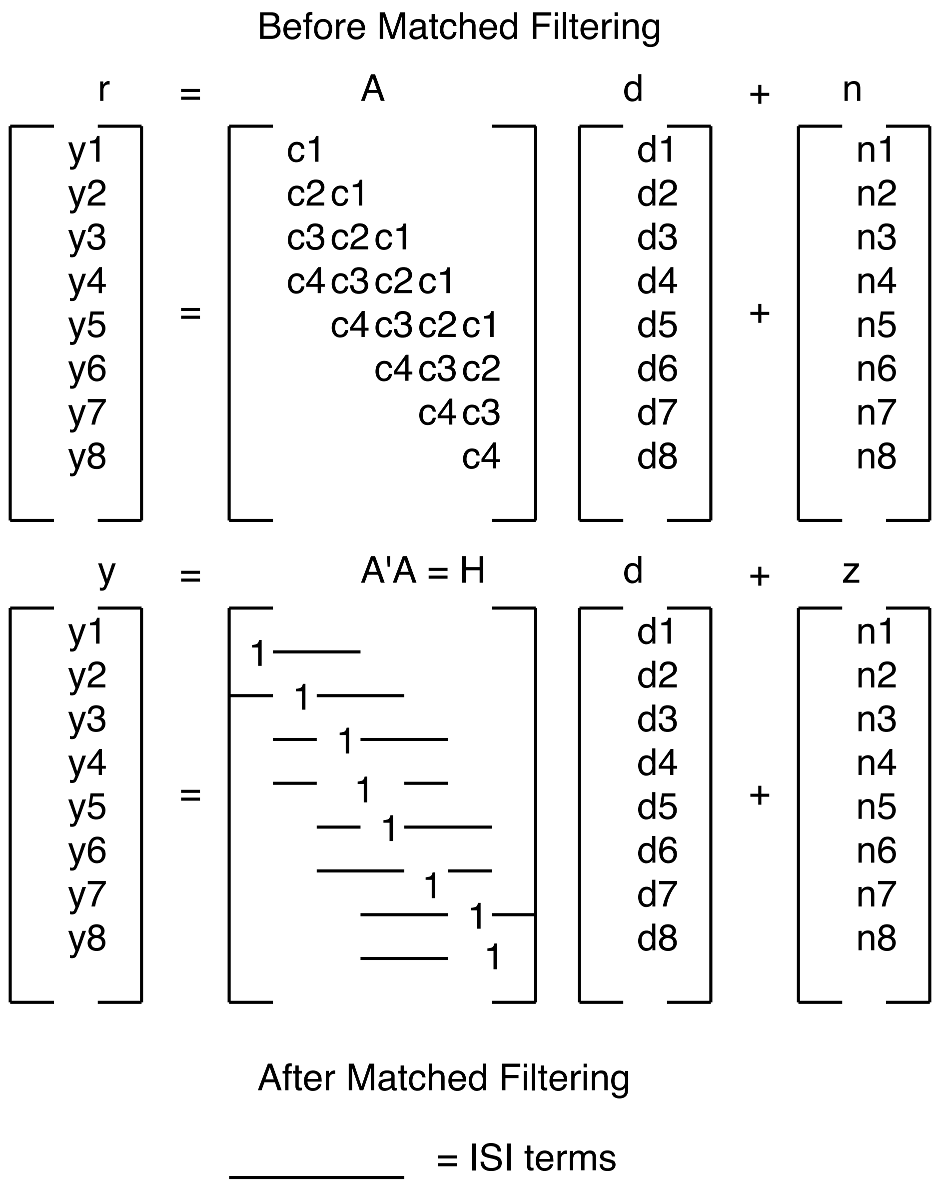


- Consists of a delay line with multiple taps
- Single/Multiple Inputs, Single/Multiple Outputs
- Generalised Channel Description $y = Ad + n$
- Matrix A describes channel

Generalised Channel Modelling

- Coefficients $c_1, \ldots, c_4$ represent channel coefficients
- $\mathbf{d}$ is data vector, $\mathbf{n}$ is noise vector
- Matrix H describes input-output relationship
- In Linear Algebra the despreading function (Matched Filter output) is

$$
\begin{aligned}
\mathbf{y} &= \mathbf{A}_r^T \mathbf{A} \mathbf{d} + \mathbf{A}_r^T \mathbf{n} \\
&= \mathbf{H} \mathbf{d} + \mathbf{z}
\end{aligned}
$$



Channel Modelling (1) - Single User

$$\mathbf{A} = \begin{bmatrix} s_{1,1} & & & \\ s_{1,2} & & & \\ \vdots & & & \\ s_{1,N} & & & \\ & s_{2,1} & & \\ & s_{2,2} & & \\ & \vdots & & \\ & s_{2,N} & & \\ & & \ddots & s_{T_u,1} \\ & & & s_{T_u,2} \\ & & & \vdots \\ & & & s_{T_u,N} \end{bmatrix}$$

- Single User A Matrix
- Spreading Code is N chips long
- T_u time samples
- Could also represent Steering vectors from an antenna array

Channel Modelling (2) - Multiple Users - Synchronous

$$\mathbf{A} = \begin{bmatrix} s_{1,1}^{(1)} & \cdots & s_{1,1}^{(K)} & & & & & & \\ s_{1,2}^{(1)} & \cdots & s_{1,2}^{(K)} & & & & & & \\ \vdots & & \vdots & & & & & & \\ s_{1,N}^{(1)} & \cdots & s_{1,N}^{(K)} & & & & & & \\ & & & s_{2,1}^{(1)} & \cdots & s_{2,1}^{(K)} & & & \\ & & & s_{2,2}^{(1)} & \cdots & s_{2,2}^{(K)} & & & \\ & & & \vdots & & \vdots & & & \\ & & & s_{2,N}^{(1)} & \cdots & s_{2,N}^{(K)} & & & \\ & & & & & & s_{Tu,1}^{(1)} & \cdots & s_{Tu,1}^{(K)} \\ & & & & & & s_{Tu,2}^{(1)} & \cdots & s_{Tu,2}^{(K)} \\ & & & & & & \vdots & & \vdots \\ & & & & & & s_{Tu,N}^{(1)} & \cdots & s_{Tu,N}^{(K)} \end{bmatrix}$$

- Multiple User-Synchronous Symbol A Matrix
- Spreading Code is N chips long
- T_u time samples
- K users present

Channel Modelling (3) - Multiple Users - Asynchronous

$$
\mathbf{A} = \begin{bmatrix}
s^{(1)}_{1,1} & & & & & & & & & \\
s^{(1)}_{1,2} & \ddots & s^{(K)}_{1,1} & & & & & & & \\
\vdots & & s^{(K)}_{1,2} & & & & & & & \\
s^{(1)}_{1,N} & & \vdots & s^{(1)}_{2,1} & \ddots & & & & & \\
 & & s^{(K)}_{1,N} & s^{(1)}_{2,2} & & s^{(K)}_{2,1} & & & & \\
 & & & \vdots & & s^{(K)}_{2,2} & & & & \\
 & & & s^{(1)}_{2,N} & & \vdots & \ddots & & & \\
 & & & & & s^{(K)}_{2,N} & & s^{(1)}_{T_u,1} & \ddots & \\
 & & & & & & & s^{(1)}_{T_u,2} & & s^{(K)}_{T_u,1} \\
 & & & & & & & \vdots & & s^{(K)}_{T_u,2} \\
 & & & & & & & s^{(1)}_{T_u,N} & & \vdots \\
 & & & & & & & & & s^{(K)}_{T_u,N}
\end{bmatrix}
$$

- Multiple User-Asynchronous Symbol A Matrix
- Fixed delays between users, time ordered
- Sync. Chip System
- Spreading Code is N chips long
- T_u time samples
- K users present

Channel Modelling (4) - Conventional DS/CDMA Single User Detector

- First Term is signal of interest, Second term in interference
- Conventional Receiver ignores the MAI and treats it as noise
- The magnitude of the interference depends on:-
  - The sign of the data
  - The cross-correlation of the spreading codes
  - the number of interfering users and their power level

$$
\begin{aligned}
y^{(k)} &= \sum_{t_c=1}^{N} e_{t_c}^{(k)} s_{t_c}^{(k)} + \sum_{\substack{k'=1 \\ (k' \neq k)}}^{K} \sum_{t_c=1}^{N} e_{t_c}^{(k)} s_{t_c}^{(k')} \\
&= \sum_{t_c=1}^{N} s_{t_c}^{(k)} s_{t_c}^{(k)} d^{(k)} + \sum_{\substack{k'=1 \\ (k' \neq k)}}^{K} \sum_{t_c=1}^{N} s_{t_c}^{(k)} s_{t_c}^{(k')} d^{(k')}
\end{aligned}
$$

Channel Modelling (5) - Async/Sync Bit

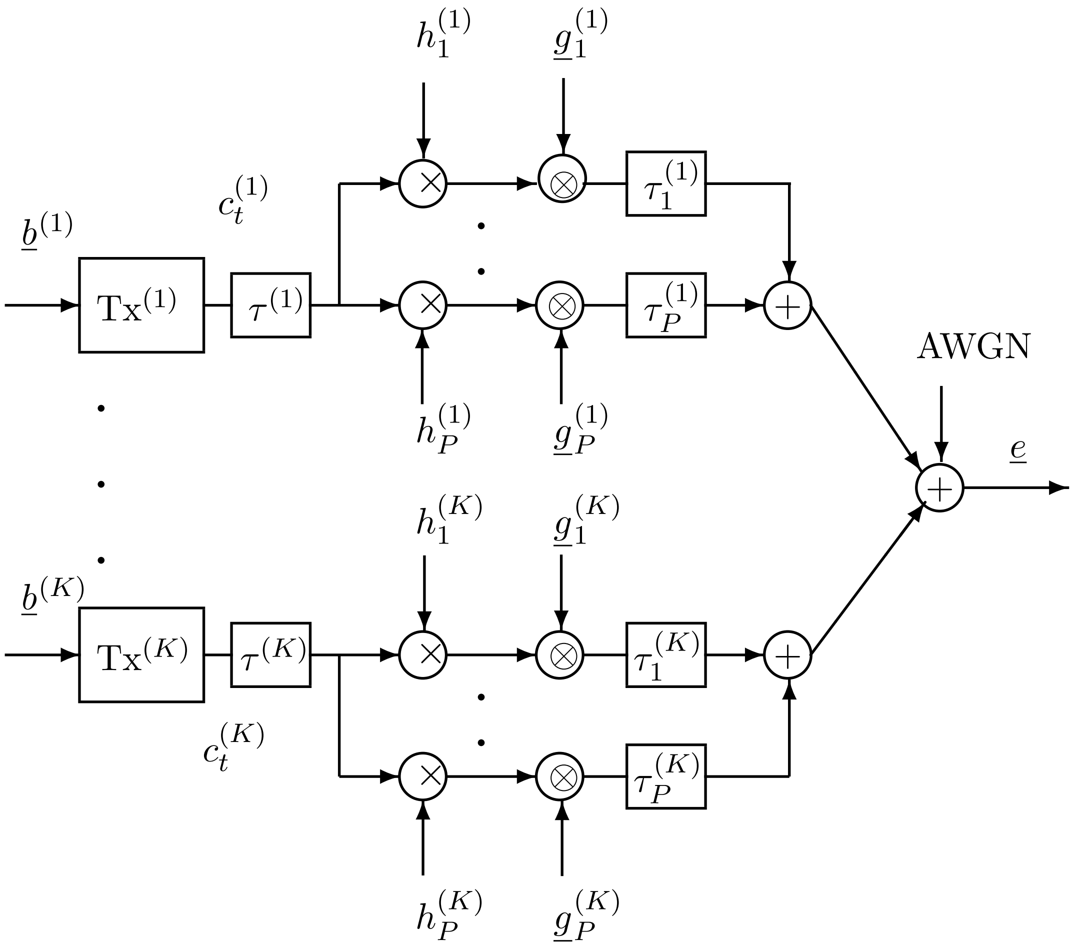
For the asynchronous symbol case an example matrix $\mathbf{H}$ is

$$
\begin{bmatrix}
h_{1,1} & \cdots & h_{1,1+K-1} & & & & \\
h_{2,1} & h_{2,2} & \cdots & h_{2,2+K-1} & & & \\
\vdots & & \ddots & & & & \\
h_{K,1} & \cdots & h_{K,K} & \cdots & h_{K,K+K-1} & & \\
 & h_{K+1,2} & \cdots & h_{K+1,K+1} & \cdots & h_{K+1,2K} & \\
 & & & \ddots & \ddots & \ddots & \\
 & & & h_{KT_u,KT_u-(K-1)} & \cdots & h_{KT_u,KT_u-1} & h_{KT_u,KT_u}
\end{bmatrix}
$$

For the symbol synchronous case the matrix structure $\mathbf{H}$ is

$$
\begin{bmatrix}
h_{1,1} & \cdots & h_{1,K} & & & & & & & \\
\vdots & & \vdots & & & & & & & \\
h_{K,1} & \cdots & h_{K,K} & & & & & & & \\
 & & & h_{K+1,K+1} & \cdots & h_{K+1,2K} & & & & \\
 & & & \vdots & & \vdots & & & & \\
 & & & h_{2K,K+1} & \cdots & h_{2K,2K} & & & & \\
 & & & & & & \ddots & & & \\
 & & & & & & & h_{KT_u-K,KT_u-K+1} & \cdots & h_{KT_u-K+1,KT_u} \\
 & & & & & & & \vdots & & \vdots \\
 & & & & & & & h_{KT_u,KT_u-K+1} & \cdots & h_{KT_u,KT_u}
\end{bmatrix}
$$

Modelling of DS/CDMA, Antenna Arrays, and Multipath



- Shows channel model for DS/CDMA, multi-path and antenna array reception
- $\otimes$ indicates the Kronecker product
- b represents infomation bits
- h represents multipath channel co-efficients
- g represents steering vector of antenna array
- τ represents multipath delay

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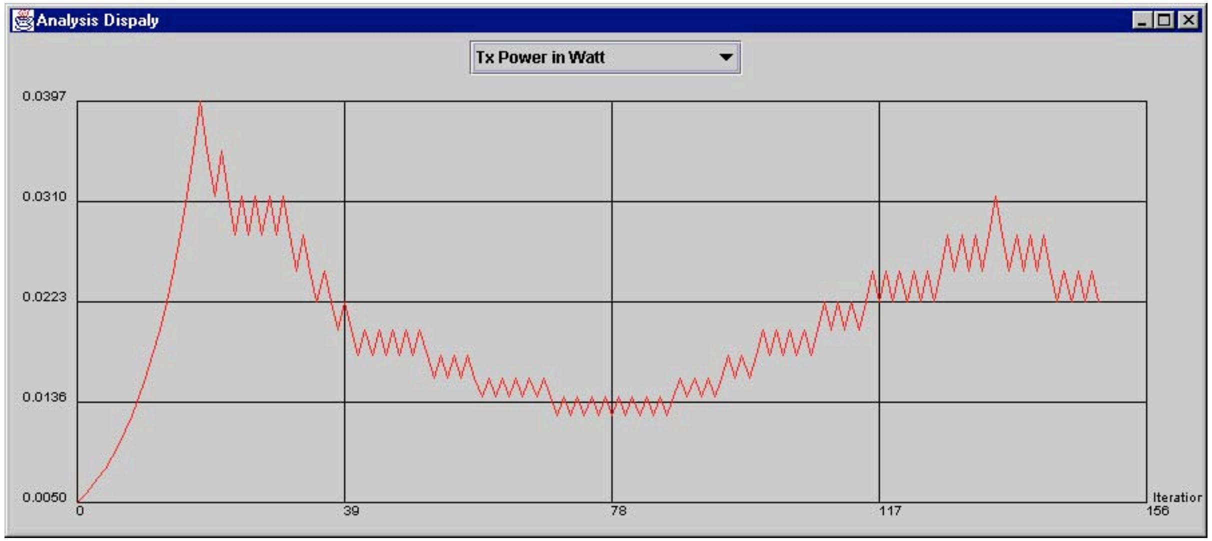
Maximising Capacity in Multi-access Cellular Systems

Minimise interference with:-

- Power Control (Tight and Loose)
  - Tight for DS/CDMA Systems
  - Loose for GSM (2G) Systems
- Handover between Cells (Soft and Hard)
  - Soft for DS/CDMA Systems
  - Hard for GSM (2G) like Systems
- Network Design and Layout
  - Basestation Position
  - Coverage/Capacity Maximisation
  - Minimise initial cost of a network
- Improved Baseband Signal Processing Algorithms

Power Control



- Near-Far problem is when a user close to base station “drowns” out a user far away from the basestation

- Power control tries to control power levels such that at the receiver all users are received with the same power

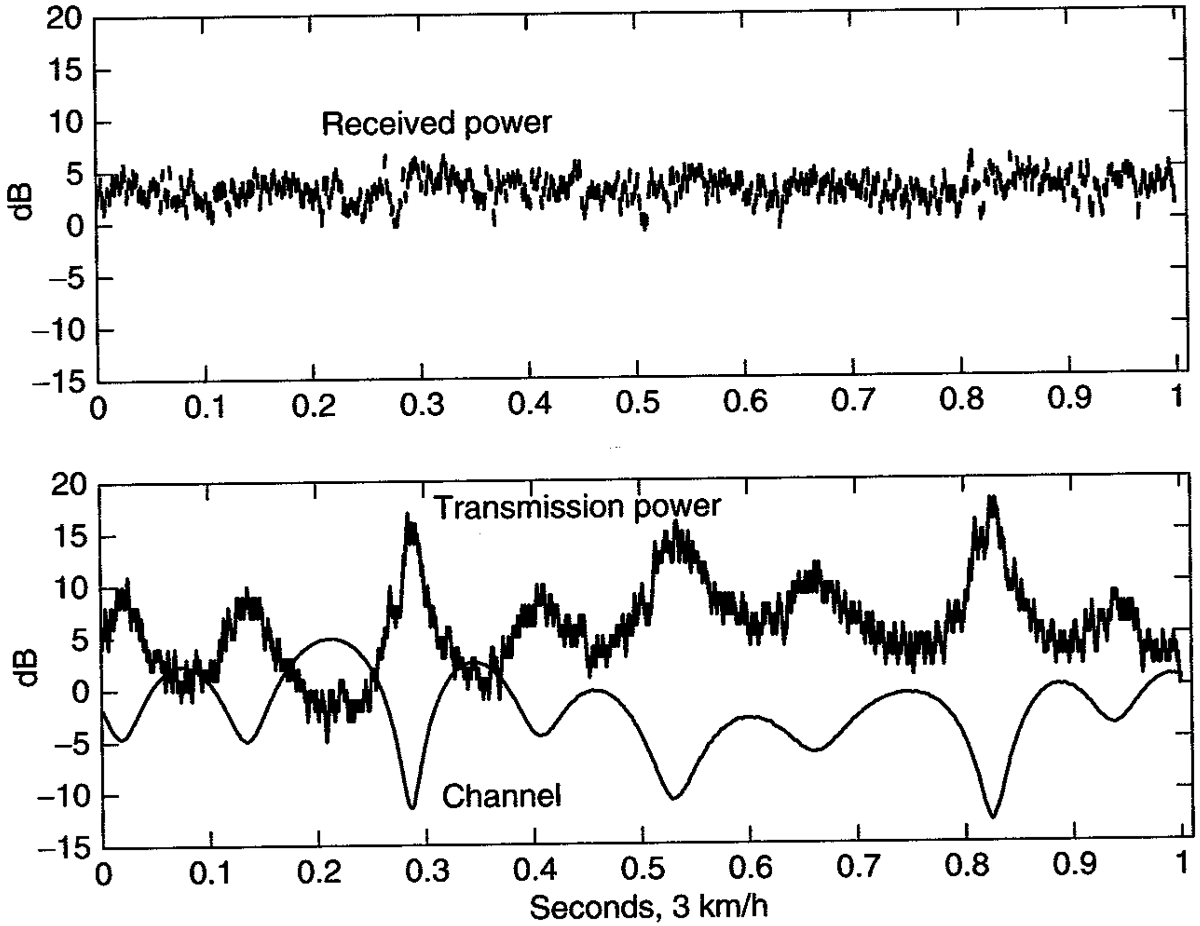
- This is difficult with dynamic systems such as mobile cellular (mobility)

- This is more critical in DS/CDMA as interference is Co-channel

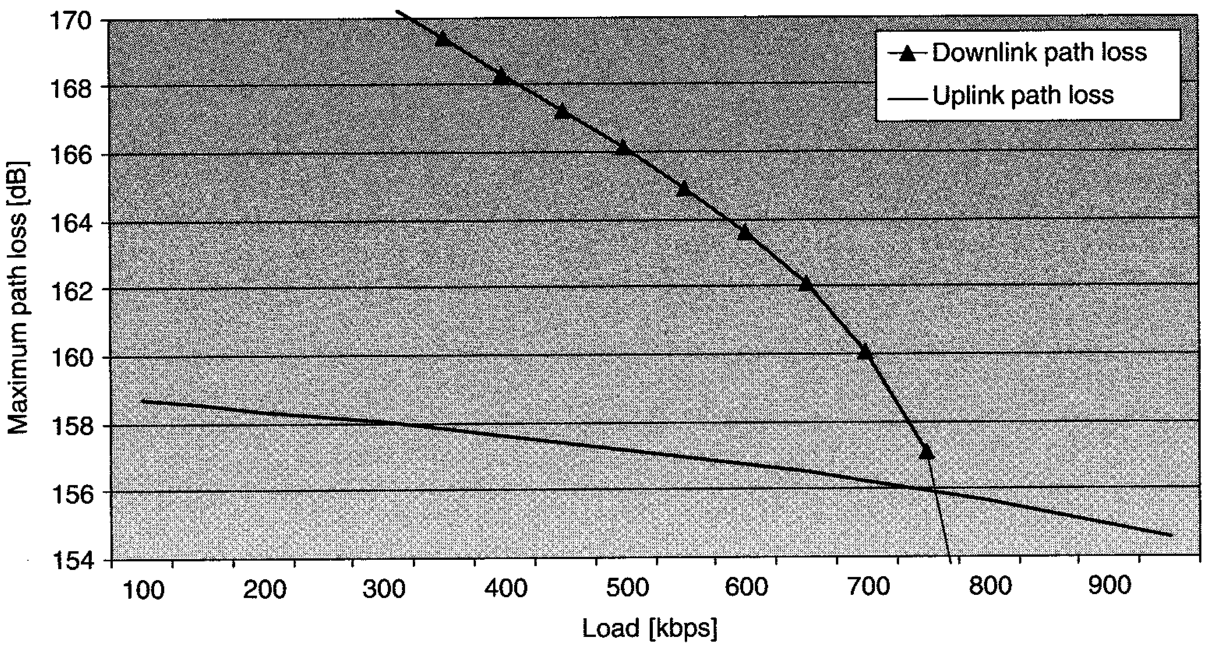
- This is still needed in GSM (narrowband) systems to minimise ACI

Power Control in Action

- Figure shows power control algorithm tracking channel (inverse)



Coverage vs. Capacity in Up/downlinks



- The uplink is coverage limited
- The downlink is Capacity limited
- This result is important in the selection of receiver algorithms
- ie. no point selecting a capacity improving algorithm if the problem is coverage!!

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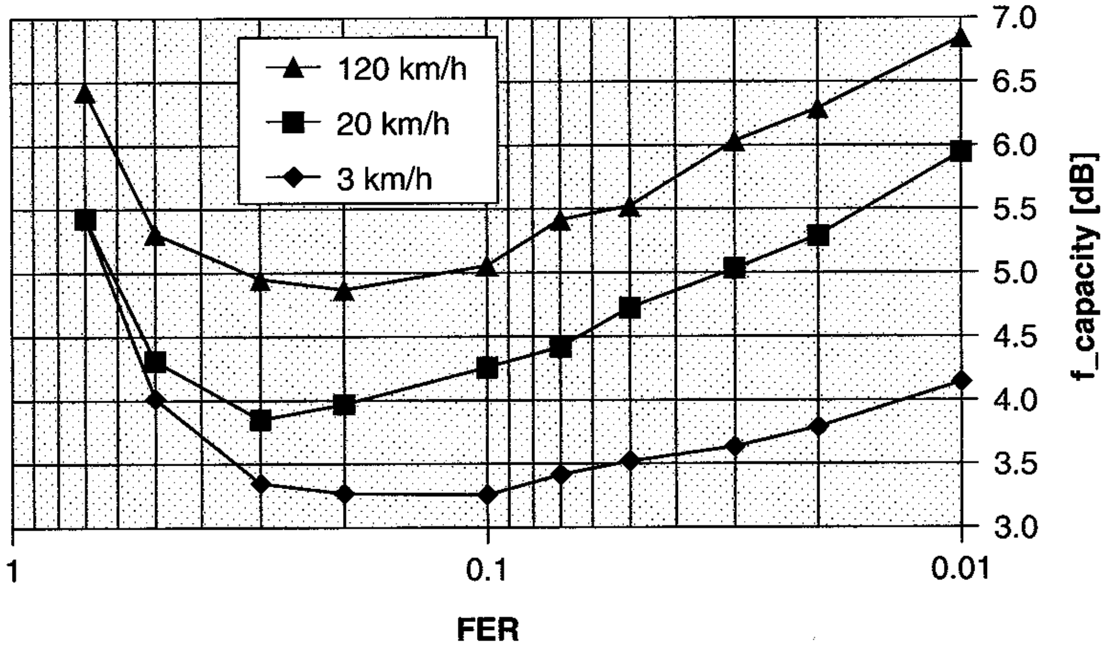
FER vs. Throughput

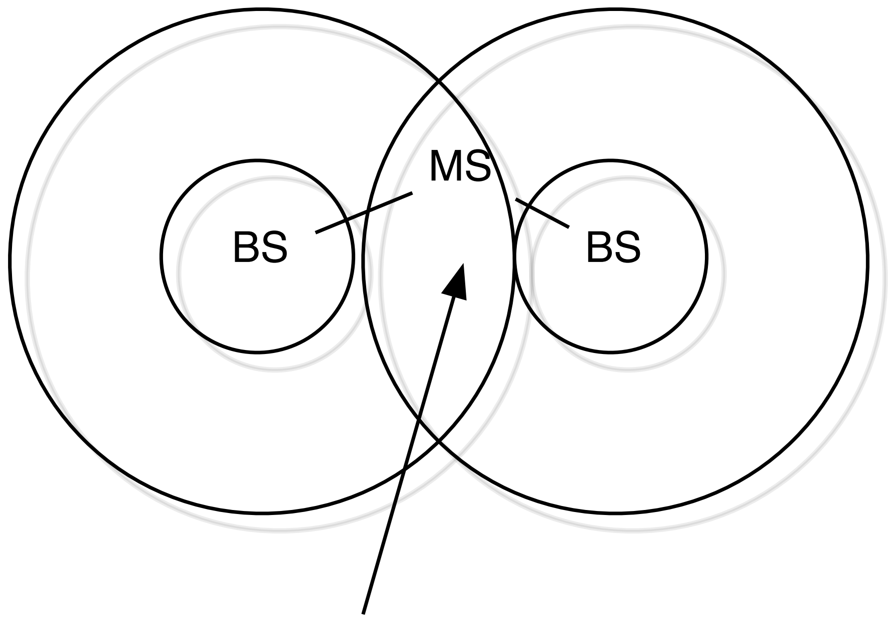
- Trade-off:
- - The Lower the FER the more re-transmits
- - The higher the FER the less users can be supported
- What is the FER to maximise system throughput?



- This result is for 32kbps and says the FER should equal 10^{-1}, roughly the Bit Error Rate should be between 10^{-2} and 10^{-3} (very low!!)

Handover between Cells - 20-40% of connections

- Soft Handover means “make before break” connection
- Hard Handover would be “break before make” connection
- Figure shows soft handover region between two cells
- The soft handover region depends on system configuration.
- soft/softer handovers are an essential interference mitigating tool



- Communication to two basestations concurrently
- Softer handover means RAKE combining of the signals occur
- Soft handover means a selection between the two signals is made
- Two power control loops are active in soft handover, additional resources are needed, rake receiver channels in BS and rake fingers in the MS

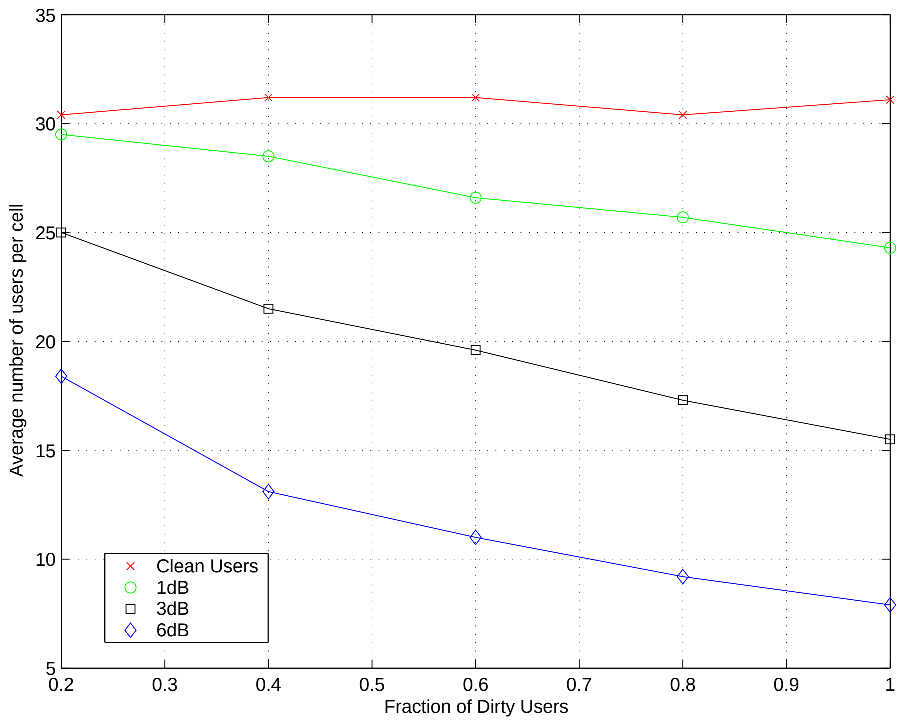
Network Design and Layout

- Expensive optimisation tools can be used to determine position of Basestations

- Statistical tools can be used to determine capacity of network

- Tools can save millions of dollars in infrastructure costs

- Tools can be programmed with different algorithms to show effect on network



Example, Network capacity if bad (dirty) users require extra power. Here capacity of network is determined when the percentage of dirty users increases

Baseband Signal Processing Algorithms

- Improved coding techniques (for noise limited systems)
- Improved interference cancellation techniques (for interference limited systems)
- Multiuser detection techniques for DS/CDMA treat users together not separately
- Better understanding of the channel helps in designing better receivers

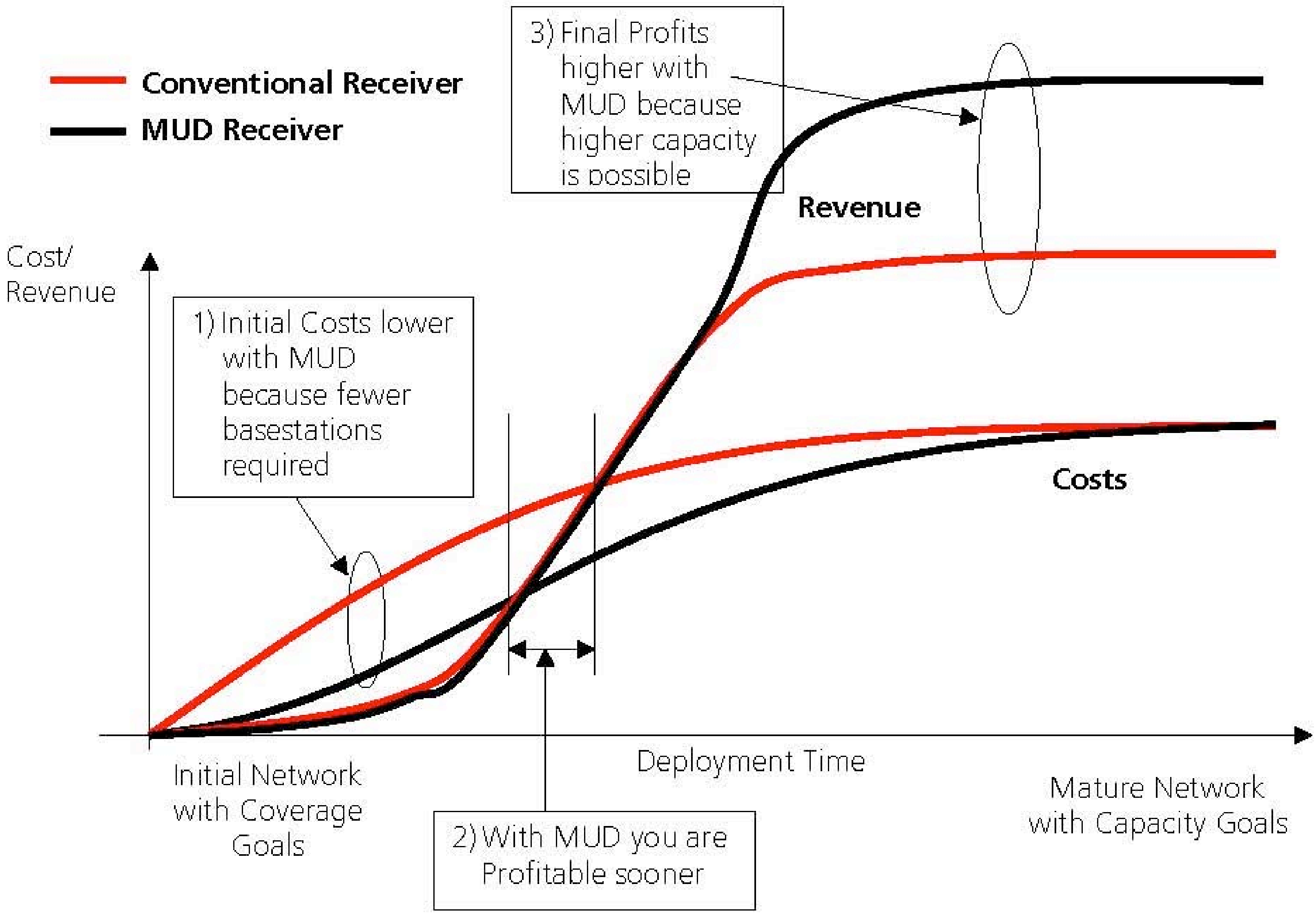
Cellular Network Life Cycle (1)

- During initial deployment the service provider is interested in maximising coverage. Here for a system with MUD the initial costs are lower due to a reduction in the number of basestations required (larger area covered by each basestation).

- With a MUD-enabled system the mobile network is profitable sooner as the revenue from the system increases.

- In the mature phase of the network operation the profits are substantially higher than with a conventional design as the system supports higher capacity and therefore more users and higher data rates.

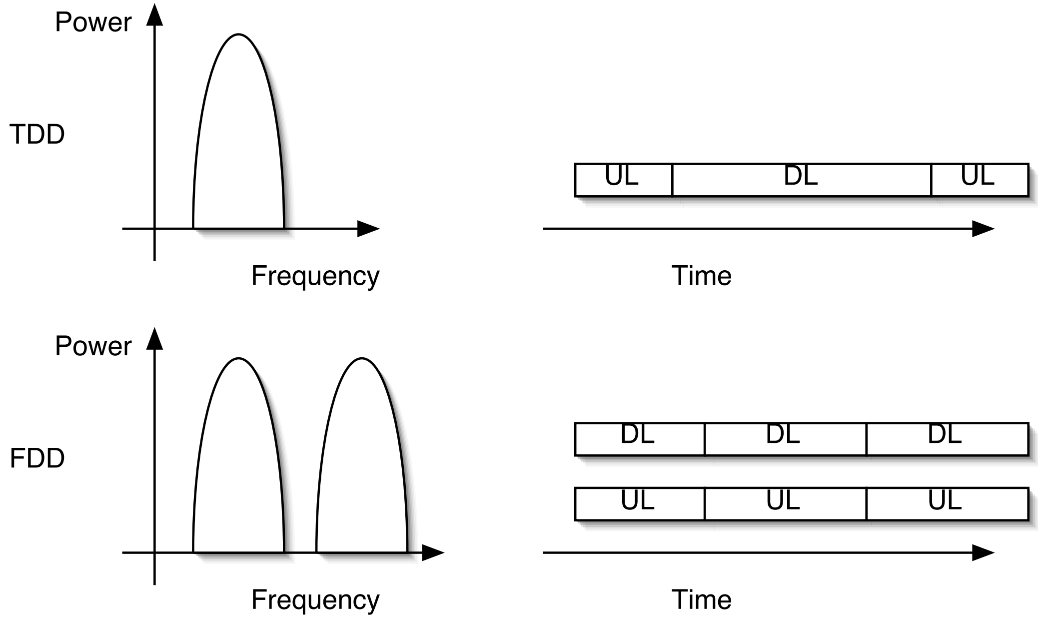
Cellular Network Life Cycle (2)



Duplex Possibilities

- Time Division Duplex (TDD)
  - Needs only one frequency range
  - Adv- Channel measurements in on downlink can be used in uplink
  - Require co-ordination and allowance for turn-around times (GSM 32km limit)

- Frequency Division Duplex (FDD)
  - Needs two frequency bands
  - Needs RF components to give good freq separation
  - Less coordination between uplink and downlink needed



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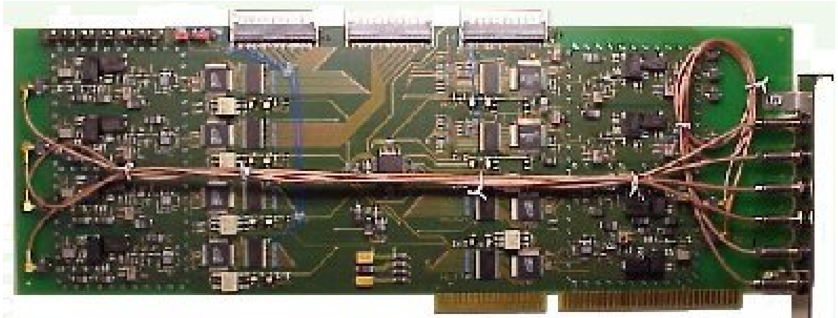
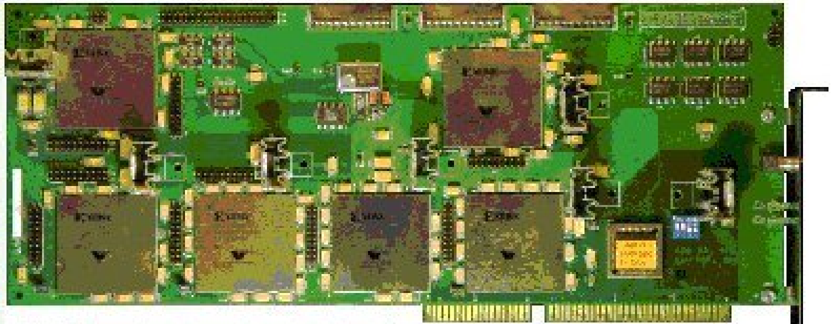
S-UMTS, Complete DS/CDMA System and Hardware Boards

Satellite - UMTS System for European Space Agency

Satellite UMTS Prototype for European Space Agency

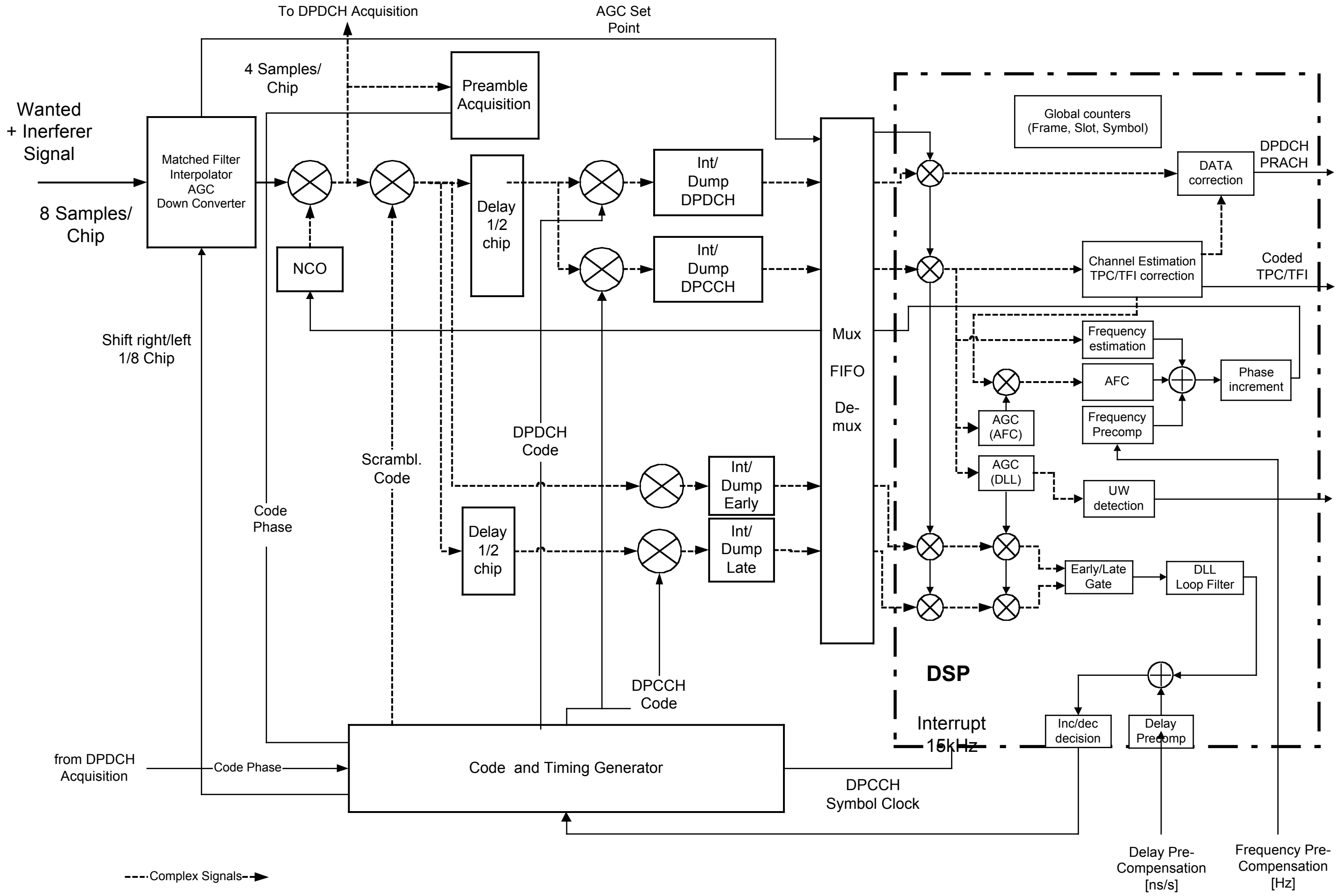


ADC board delivers data samples at 32 Mbps to the digital baseband receiver board.



S-UMTS baseband receiver circuit board incl. 6 FPGAs to acquire, synchronise, track and despread the incoming W-CDMA signal.

S-UMTS, Single User DS/CDMA Receiver Design



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S-UMTS, Research, Design, Development, and Test Flow

- Initial Specifications Determined (max frequency offset, drift, E_s/N_0)
- Design process to determine acquisition and tracking mechanism
- Simulation testing of algorithms to validate
- Conversion of algorithms to fixed point design
- Implementation of algorithms for DSP processor (C Code)
- Implementation of algorithms for FPGA processor (VHDL firmware)
- Simulation testing of algorithms, Real-time testing
- Integration and Test of DSP/FPGA/Processor
- Performance evaluation and documentation of System

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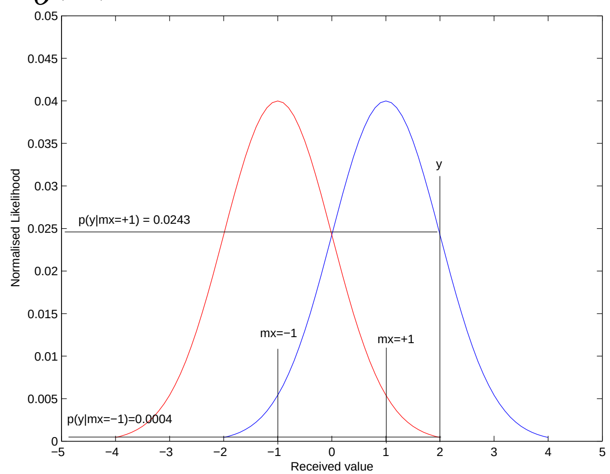
Detection for Channels without Memory

Channel Model
$$y_j = b_j + n_j$$

- A-Posteriori Prob. (APP)
$$\Pr\{b_j = b^{(m)}|y_j\} = \frac{p(y_j|b_j = b^{(m)})\Pr\{b_j = b^{(m)}\}}{p(y_j)}$$

- Maximum-A-Posteriori (MAP)
$$\arg\max_{b^{(m)}} \Pr\{b_j = b^{(m)}|y_j\} = \arg\max_{b^{(m)}} p(y_j|b_j = b^{(m)})\Pr\{b_j = b^{(m)}\}$$

- Maximum Likelihood (ML) Detection
$$\arg\max \Pr\{b_j = b^{(m)}|y_j\} = \arg\max_{b^{(m)}} p(y_j|b_j = b^{(m)})$$



- APP/MAP A-prior information provides “hint” of real data value

- $p(y_j|b_j)$ is a point from a Gaussian Distribution

Detection for Channels with Memory

The Channel Model is

$$\mathbf{y} \quad = \quad \mathbf{b} + \mathbf{n}$$

The APP outputs are

$$\Pr\{b_j = b | y_1, \cdots, y_{j+M}\}.$$

The MAP decision criterion is then

$$\begin{aligned} \hat{b}_j &= \arg\max_{b \in \pm 1} \Pr\{b_j = b | y_1, \cdots, y_{j+M}\} \\ &= \arg\max_{b \in \pm 1} \frac{p(y_1, \cdots, y_{j+M} | b_j = b) \Pr\{b_j = b\}}{p(y_1, \cdots, y_{j+M})} \\ &= \arg\max_{b \in \pm 1} p(y_1, \cdots, y_{j+M} | b_j = b) \Pr\{b_j = b\}. \end{aligned}$$

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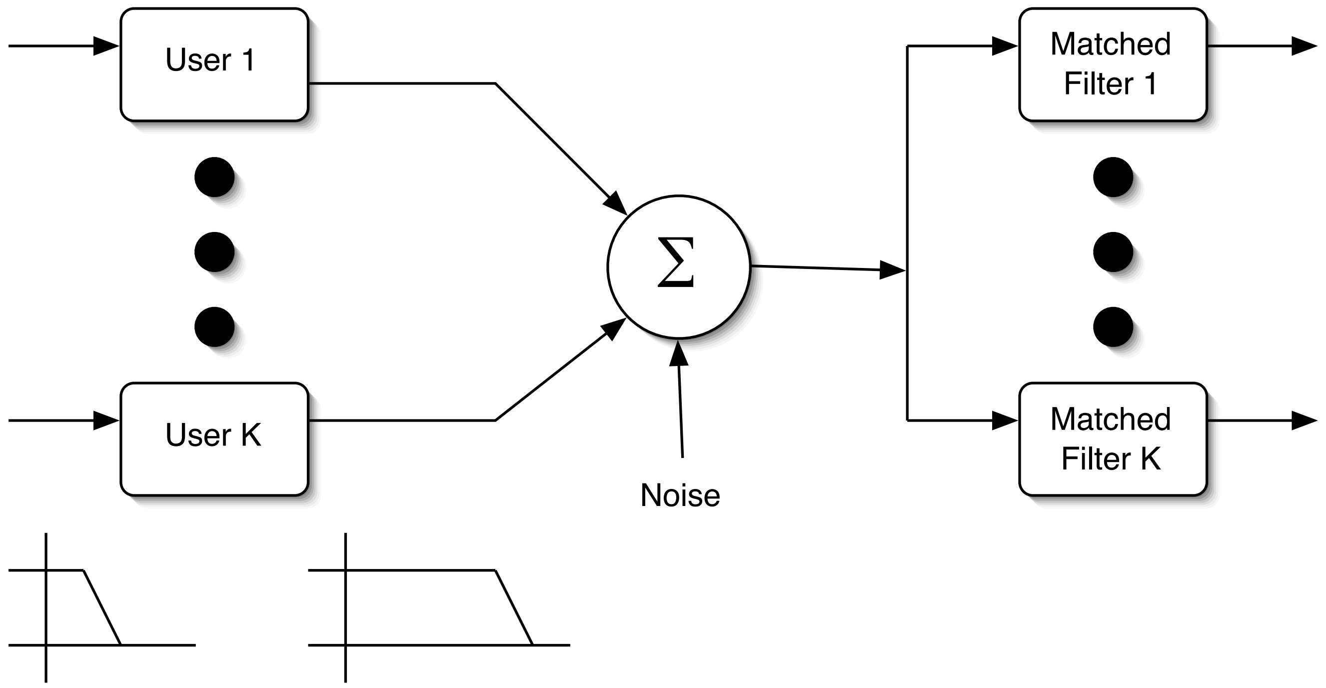
The Multiuser DS/CDMA System/Channel Model



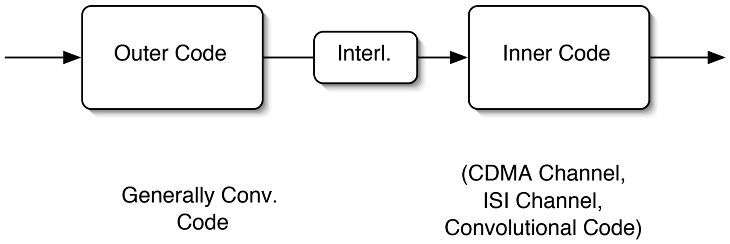
- Each User transmits with excess bandwidth (Processing Gain)
- Matched filter is optimum only when a single user is present

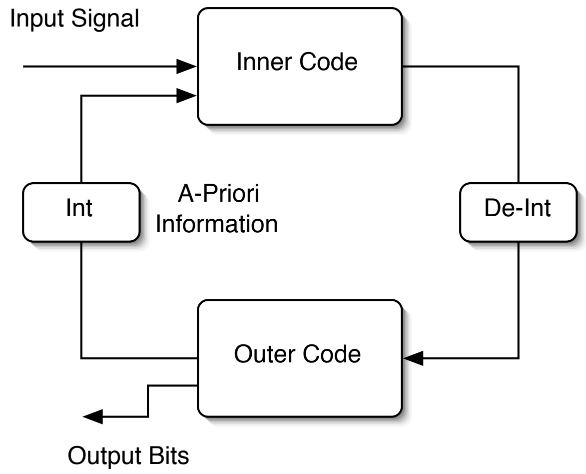
The Problem

“To Achieve Capacity for a Multiple Access Channel”

- The Optimal Solution is exponentially complex in the number of users (Verdú) (need to search every combination (000,001,010,011,100,101,110,111))
- The Traditional Solutions use :
  - Linear/Non-Linear (Decorrelator, etc.)
  - Uncoded Techniques (MMSE, etc.)
- These Solutions are suboptimal or very complex
- Ideally we need a solution that is:
  - Linear in Complexity and approaches Optimum Performance

Channel Model and Receiver Diagram

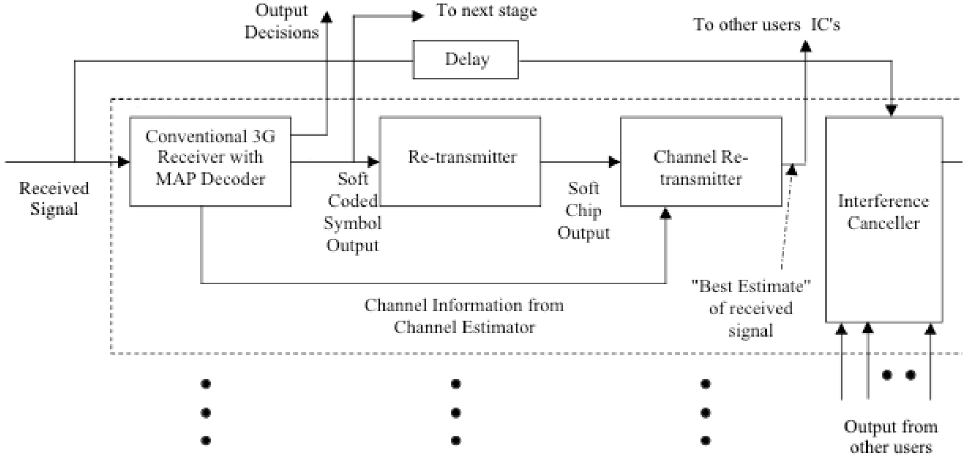


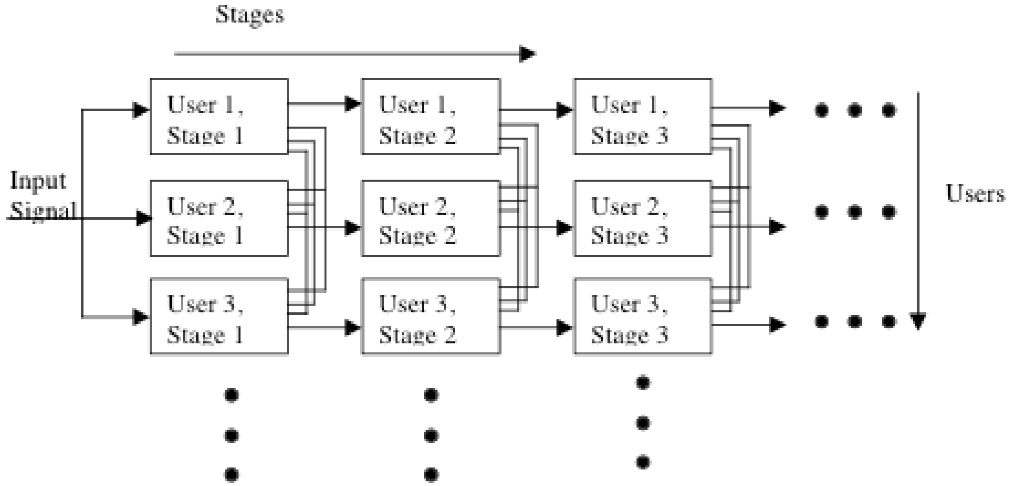


- Model consists of inner and outer code

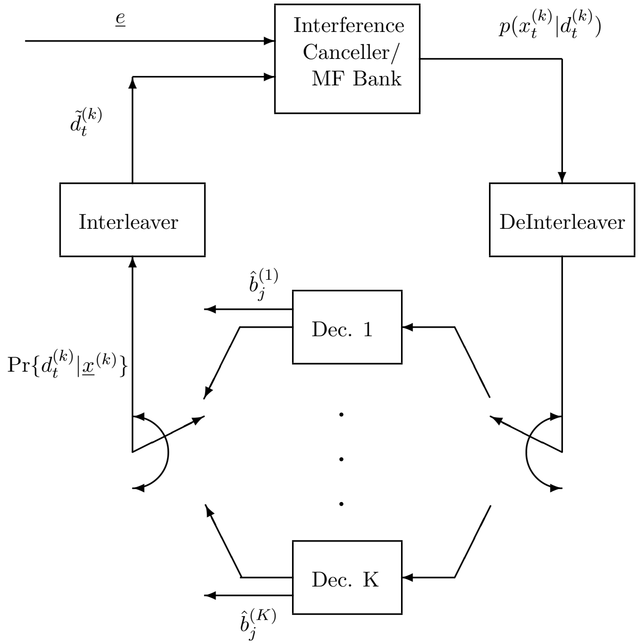
- Important is that each code is seperated by an interleaver

Turbo CDMA[1, 2]



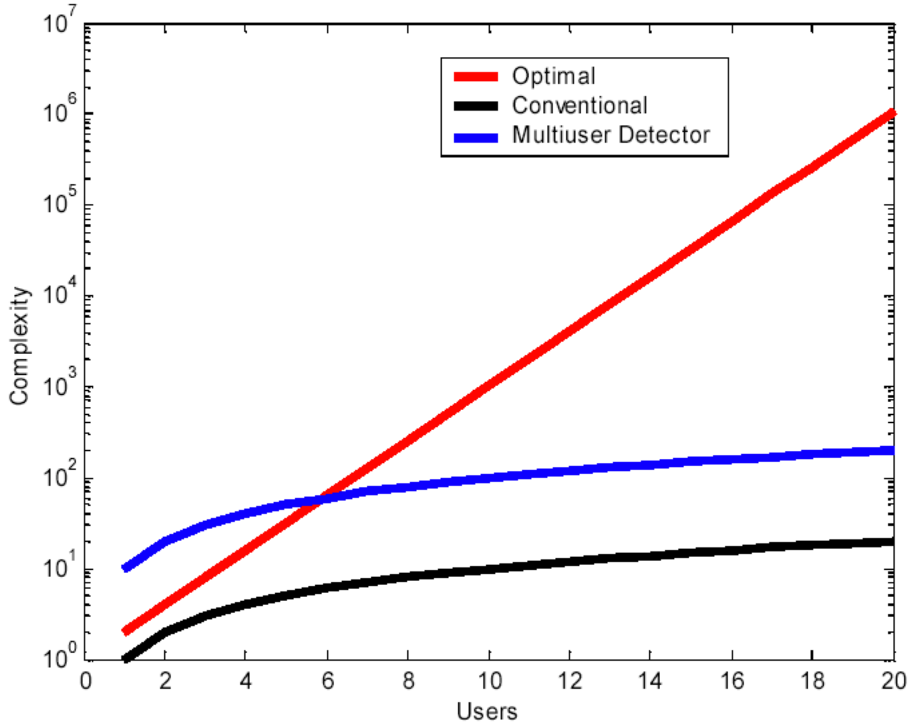


- Diagrams show realistic implementation of Iterative MUD (compare to diagram on right)
- Iterations are folded out
- Cancellation is performed at the chip level
- Same structure could be used for other “Turbo Rxs.”

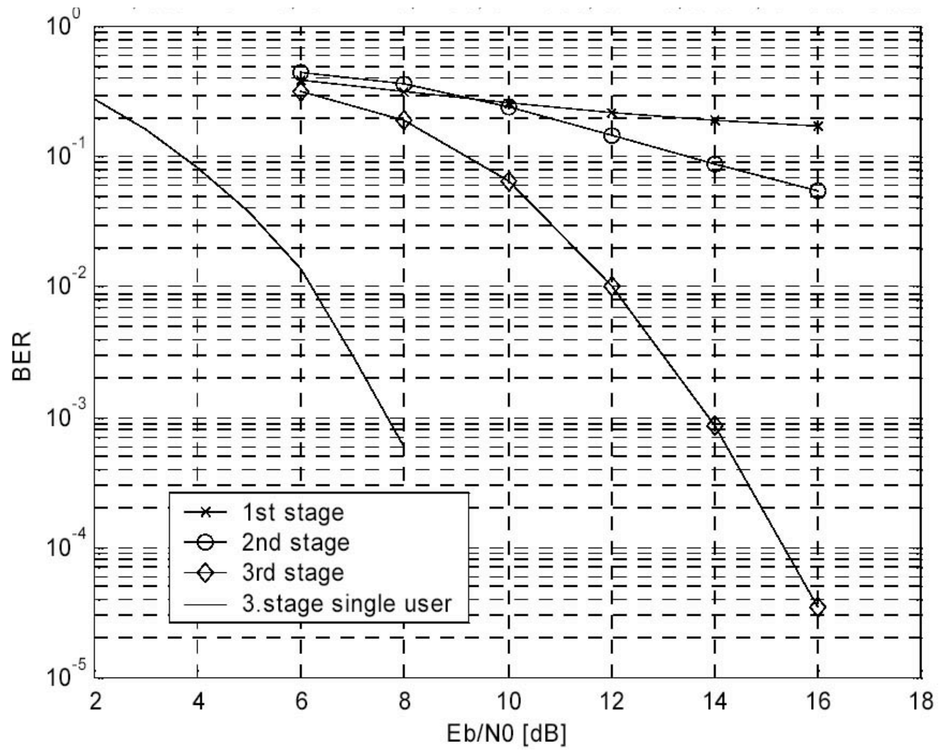


Multiuser Detection-Computational Complexity

- Complexity is approximately 10 times a Conv. Receiver
- Complexity grows on a linear scale with Users!!!
- Optimal receiver grows exponentially with the number of users
- Optimal receiver is only possible for less than 10 users
- Simulations of up to 95 users have been performed for IMUD

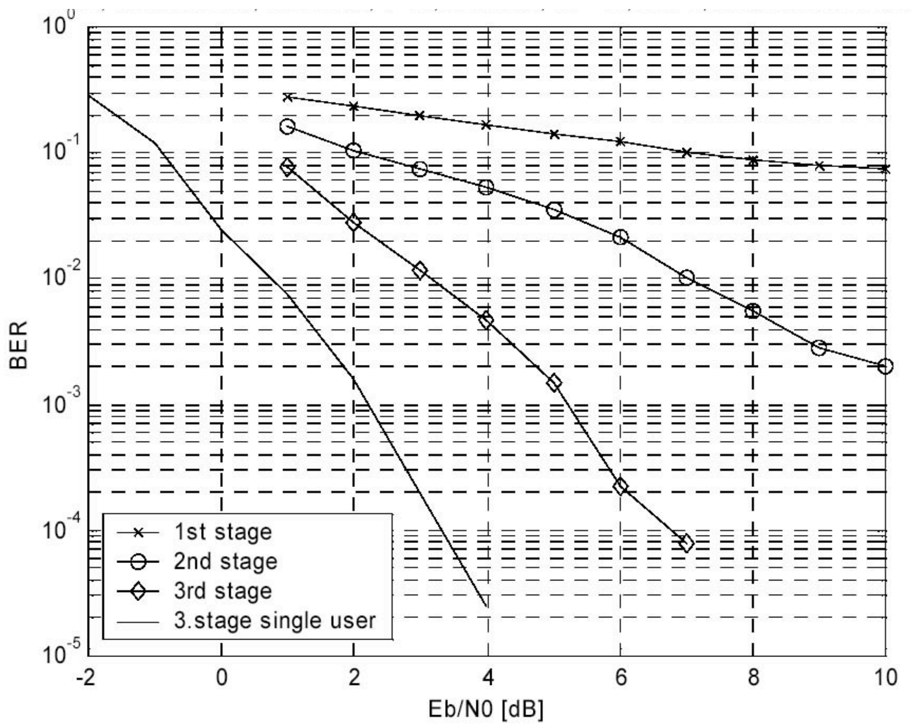


Single Cell BER Performance Results



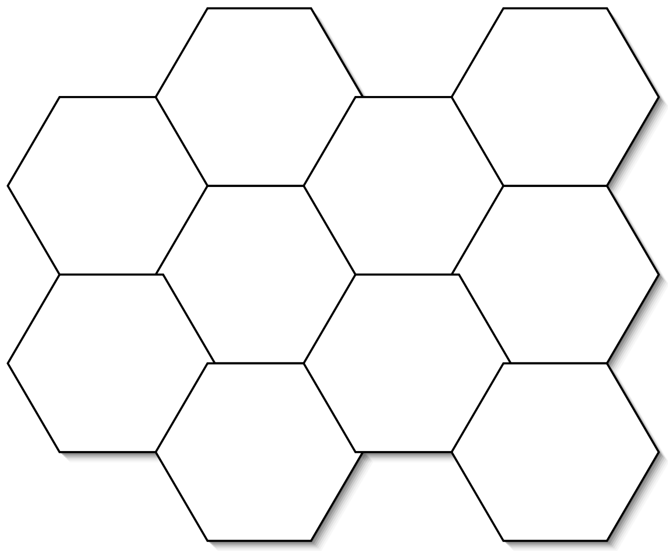
- 25 Users, Processing Gain =16
- 64 kbps per user
- Veh. A Channel, 50km/hr (Freq. Selective Fading, Fading on each Tap)
- 3GPP Release ’99 Compliant
- Single User Performance = Solid line
- Performance over Rx. iterations
- 1st Stage Perf. = Conv. Rx.
- 3rd Stage Perf. = Multiuser Rx.

Single Cell BER Performance Results - 4 Antennas[2]



- 70 Users, Processing Gain =16
- 64 kbps per user. 4 Antennas (ULA)
- Veh. A Channel, 50km/hr (Freq. Selective Fading, Fading on each Tap)
- 3GPP Release '99 Compliant
- Single User Performance = Solid line
- Performance over Rx. iterations
- 1st Stage Perf. = Conv. Rx.
- 3rd Stage Perf. = Multiuser Rx.

What do these results mean?



- In terms of System Benefit? (Capacity-Cell Size)

- In terms of Provider Benefit? (“$$$!!”)

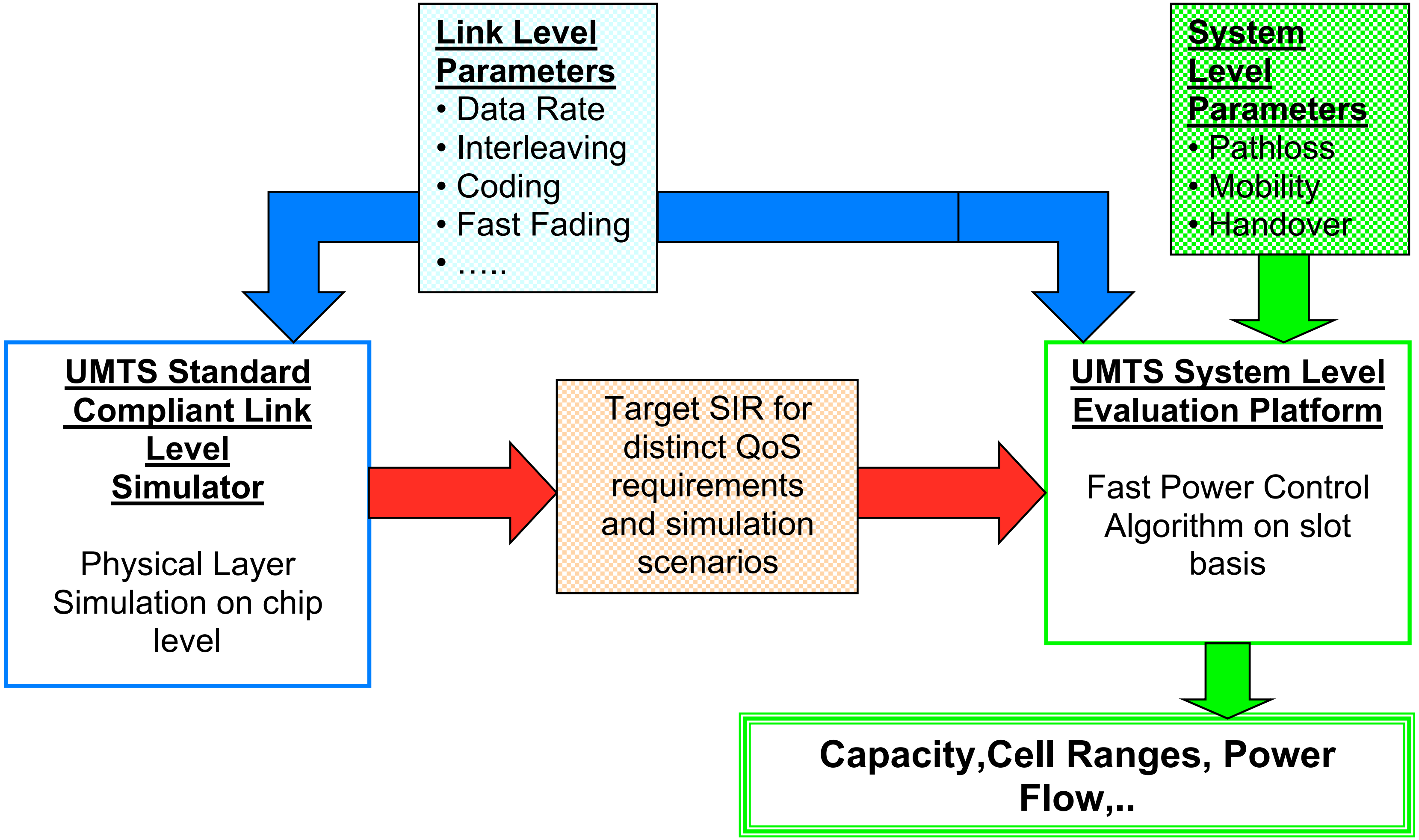
For this we need to study the system aspects!

- Multi-cell Environment

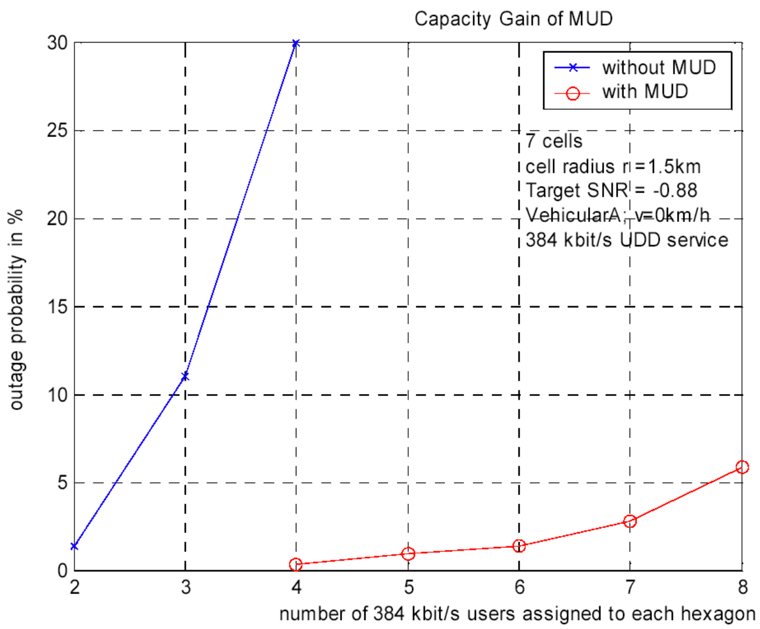
- Power Control

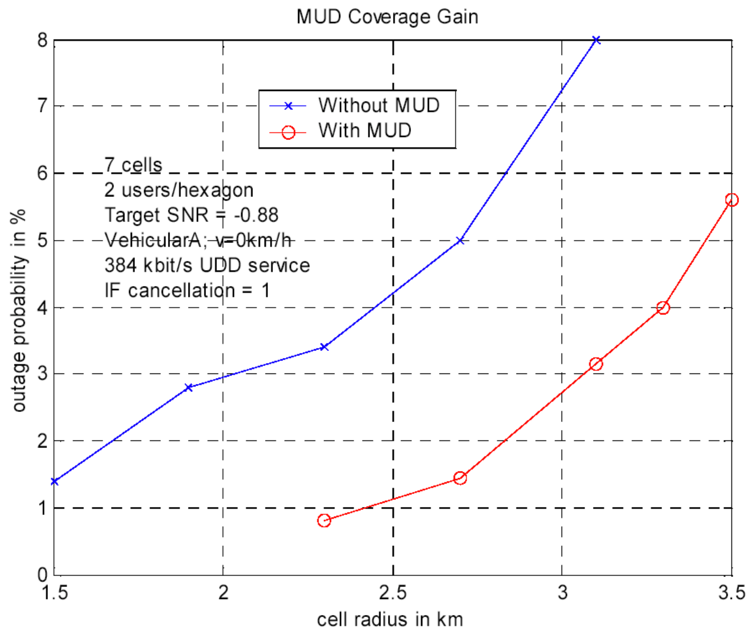
- Intra/Inter Cell Interference

Converting Link Level results to the System Level?



System Level Results





- System level tool takes into account inter/intra-cell interference
- Tool determines statistical performance
- Outage is defined as not meeting FER target
- Capacity Gain is from 2.5 to 7.5 users per cell (3 times increase)
- Coverage Gain is from 2.7 km to 3.4 km (increase of 1.5 times)

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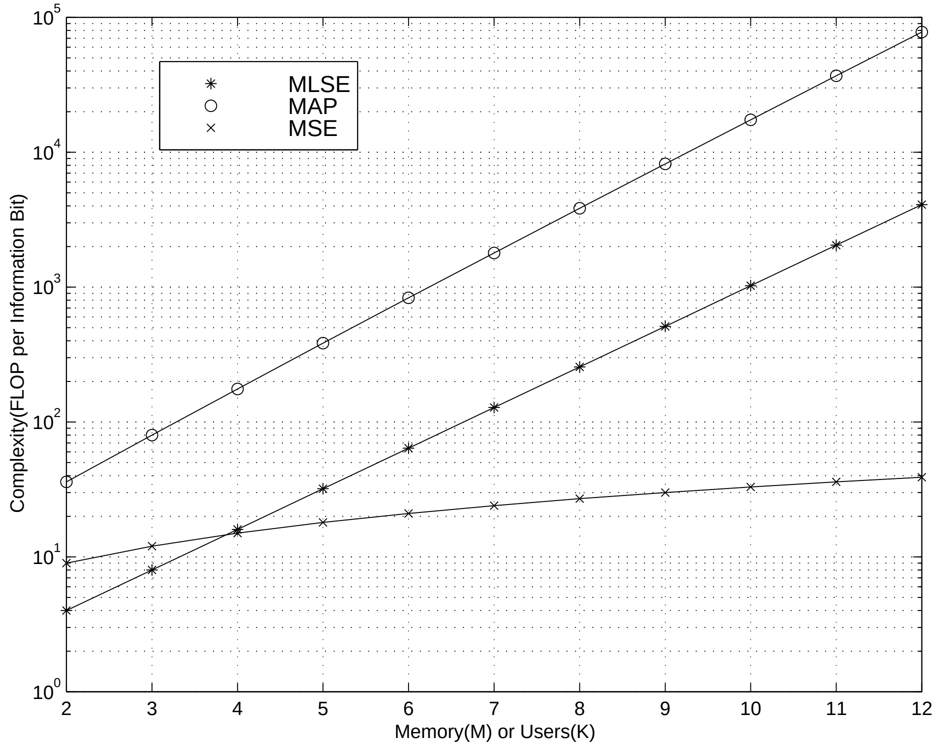
- **Interference Canceller Design**
- DS/CDMA Acquisition Problem
- MAP Algorithm

Interference Canceller - Motivation

- Complexity of MAP is exponential with Memory of Channel
- ie. $O(2^M)$, where M is the memory of channel or No. of users
- We want to reduce this complexity without compromising performance
- Ideally we want linear complexity, ie. K users = $O(K)$ Complexity



Interference Canceller-System Model [1]

We start with the familiar linear DS/CDMA channel model of

$$\mathbf{e} = \mathbf{A}\mathbf{d} + \mathbf{n}.$$

Matched Filtered DS/CDMA Channel

$$\begin{aligned} \mathbf{y} &= \mathbf{A}_r^T \mathbf{A}\mathbf{d} + \mathbf{A}_r^T \mathbf{n} \\ &= \mathbf{H}\mathbf{d} + \mathbf{z} \end{aligned}$$

We rearrange the MF-DS/CDMA channel model as

$$\mathbf{y} = \mathbf{d} + \mathbf{M}\mathbf{d} + \mathbf{z}$$

where we assume

$$\begin{aligned} \mathbf{M} &= \mathbf{H} - \text{diag}(\mathbf{H}) \\ \text{diag}(\mathbf{H}) &= \mathbf{I} \end{aligned}$$

Interference Canceller(2) - Details

The interference canceller attempts to cancel these interfering terms with an estimate of the data sent

$$\begin{aligned}\mathbf{x} &= \mathbf{y} - \mathbf{M}\tilde{\mathbf{d}} \\ &= \mathbf{I}\mathbf{d} + \mathbf{M}\mathbf{d} - \mathbf{M}\tilde{\mathbf{d}} + \mathbf{z} \\ &= \mathbf{d} + \mathbf{M}(\mathbf{d} - \tilde{\mathbf{d}}) + \mathbf{z}.\end{aligned}$$

Following de-interleaving we assume the noise is again white, therefore

$$\mathbf{x} = \mathbf{d} + \mathbf{M}(\mathbf{d} - \tilde{\mathbf{d}}) + \mathbf{n}.$$

where the total variance (σ_x^2) of $\mathbf{x}$ equals the variance from the remaining interference plus the noise statistic.

Interference Canceller(3) - Details

We therefore compute

$$p(x_t|d_t = \pm 1) \quad = \quad \frac{1}{\sqrt{2\pi\sigma_x^2}} \exp\left\{-\frac{(x_t \mp 1)^2}{2\sigma_x^2}\right\}.$$

$$\sigma_x^2 = E\left\{(x_t^{(k)} - d_t^{(k)})^2\right\},$$

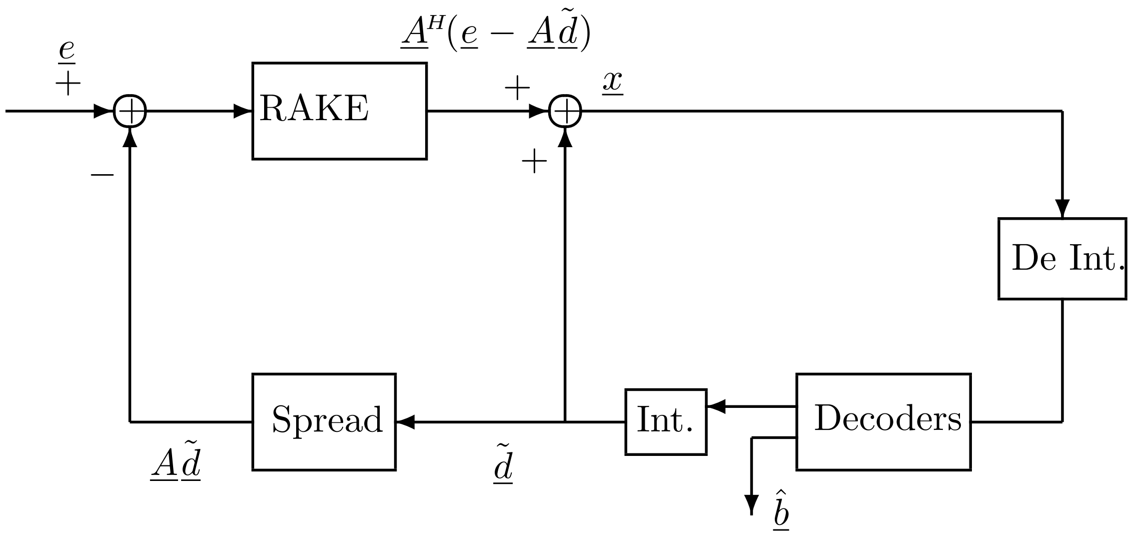
where

$$\begin{aligned}
\tilde{d}_t^{(k)} &= E\left\{d_t^{(k)}\right\} \\
&= \sum d_t^{(k)} p(d_t^{(k)}) \\
&= p(d_t^{(k)} = +1) - p(d_t^{(k)} = -1)
\end{aligned}$$

Interference Canceller (4) - Details[3, 4]

- Above method is a Symbol Level Interference Canceller
- This is still complex as M matrix needs to be computed
- The complexity can be further reduced by chip level cancellation
- Complexity is now twice a RAKE receiver (Rake function= Spread function)

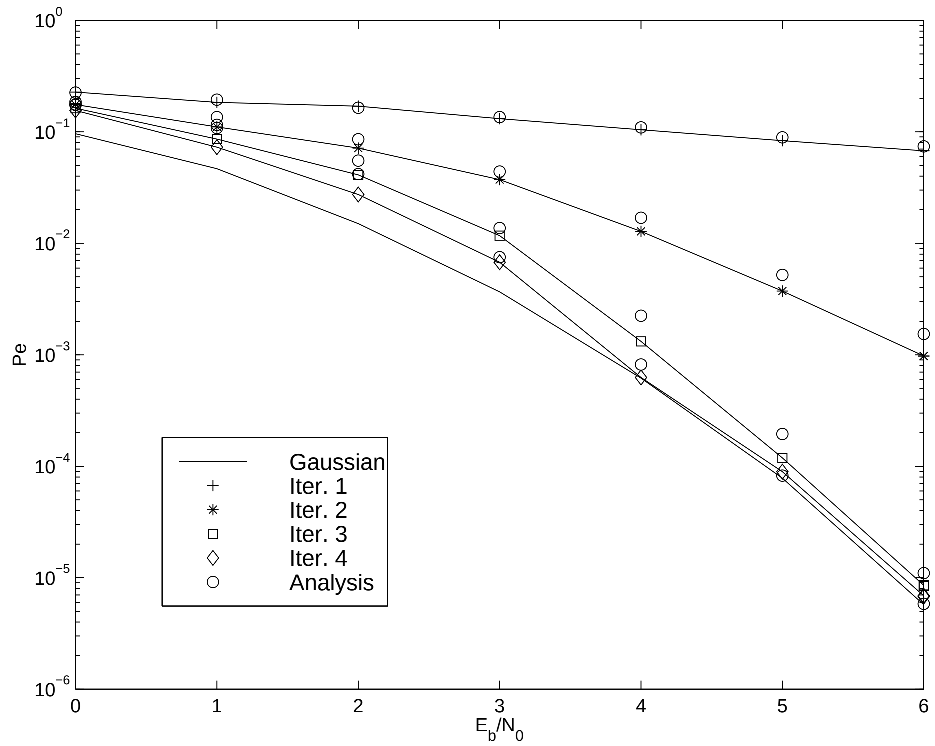
$$
\begin{aligned}
\mathbf{x} &= \mathbf{y} - \mathbf{M}\tilde{\mathbf{d}} \\
&= \mathbf{A^H}_r\mathbf{e} - (\mathbf{A^H}_r\mathbf{A}_r - \mathbf{I})\tilde{\mathbf{d}} \\
&= \mathbf{A^H}_r(\mathbf{e} - \mathbf{A}_r\tilde{\mathbf{d}}) + \mathbf{I}\tilde{\mathbf{d}}
\end{aligned}
$$



Interference Canceller - Performance

BER Performance of the Iterative Interference Canceller,

- Users ($K = 27$),
- Processing Gain (Spreading Factor) is $N = 20$.
- Random Spreading Codes
- Chip Synchronous
- Symbol Asynchronous



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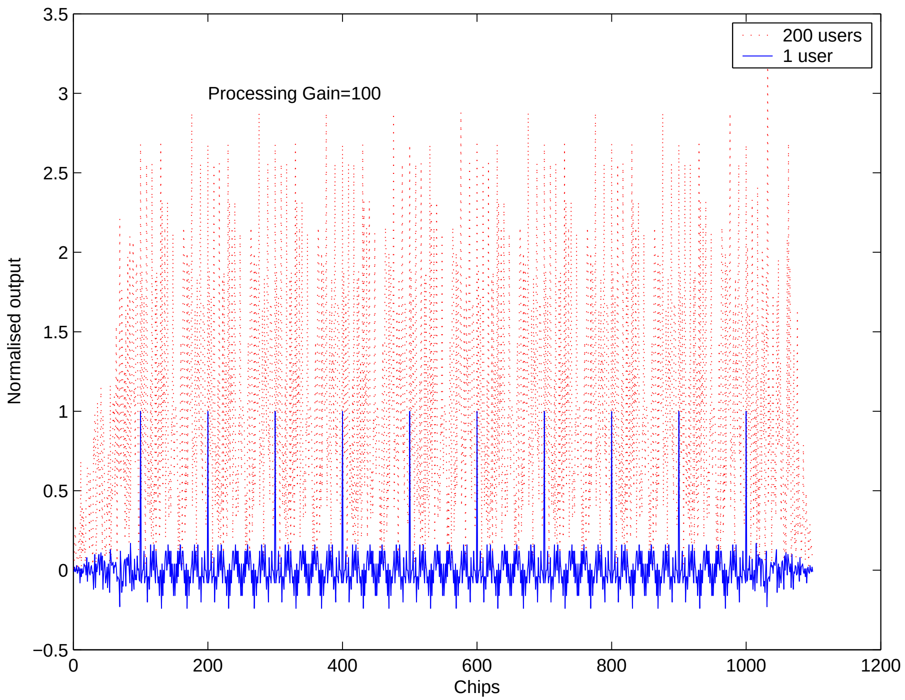
- MAP Algorithm

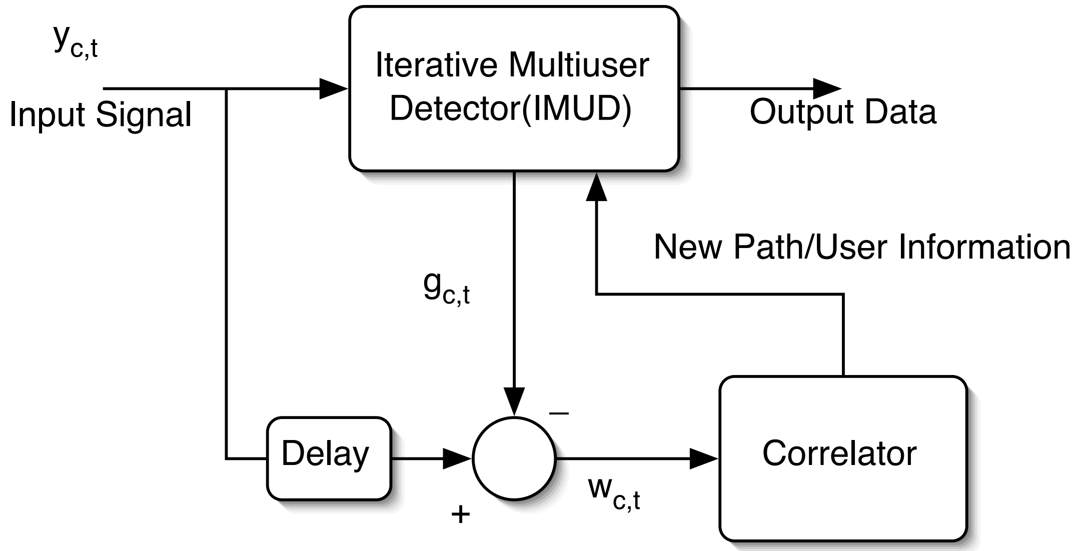
Real World Issues - The DS/CDMA Acquisition Problem[5, 6]

- Acquisition:-before signal detection/decoding, Typically using a correlator
- As the Multiple Access Interference (MAI)/Users Increases this fails as a “good” threshold point cannot be found, therefore limiting system capacity.



- Red lines show correlation output with 200 users (mainly interference)
- Blue Lines show correlation peak with 1 user (no interference)
- Determining a threshold and acquiring a signal with 200 users using a conventional correlator is impossible

Acquisition Unit Block Diagram

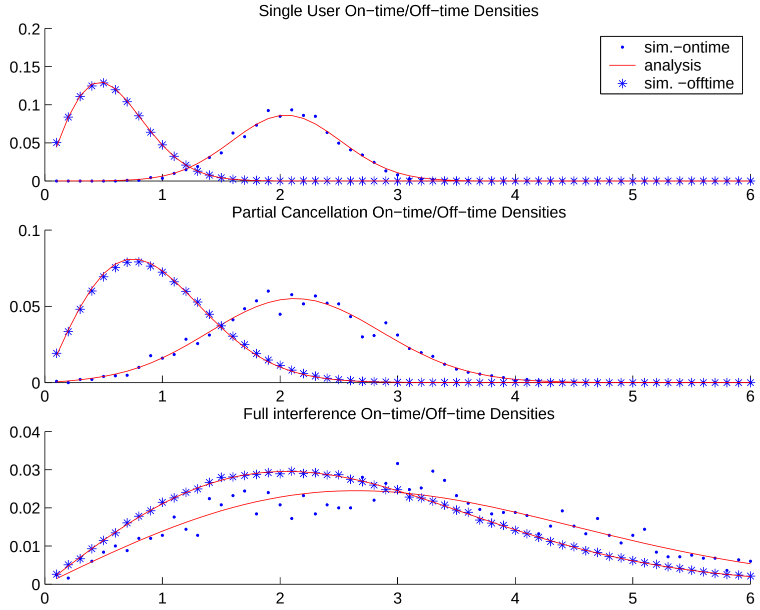


- Receiver has "good" estimate of current users, to find new users we should subtract current users i.e. $w_{c,t} = y_{c,t} - \tilde{g}_{c,t}$,

  For 1-D case, correlator performs $r_{c,t} = \left| \sum_{c=1}^{N} w_{c,t} (s_{c,-t}^{(1)})^T \right|$

Calculate Probabilities

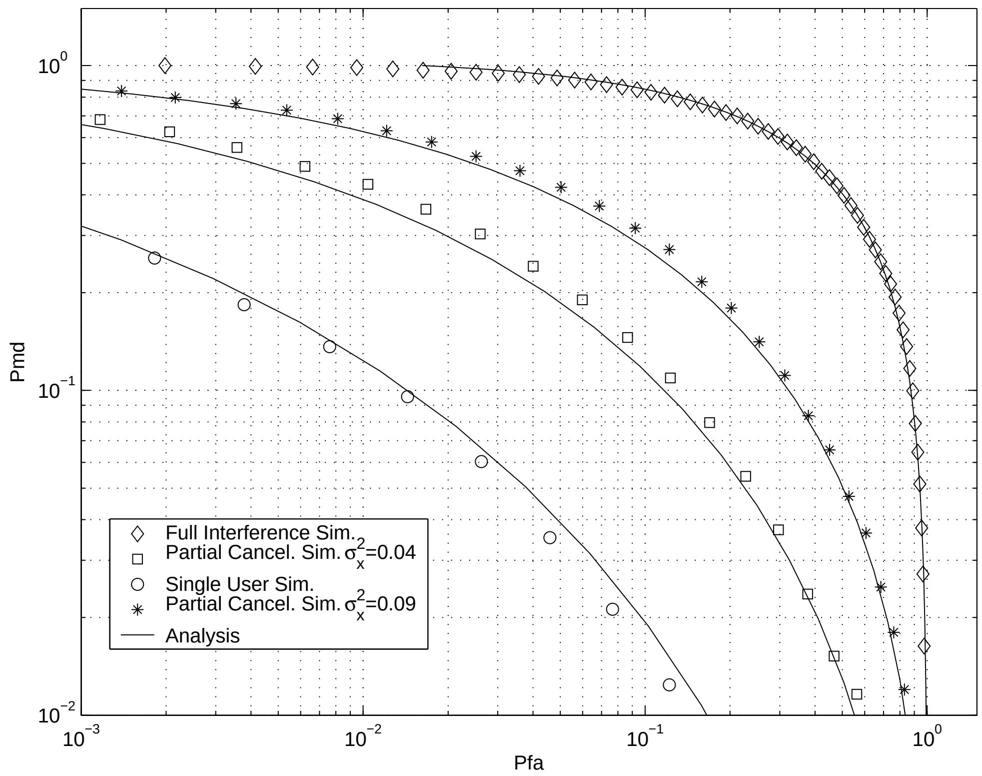
- For Analysis
  - Integrate on-time density from $0 \rightarrow r$ to determine P_{md}
  - Integrate off-time density from $r \rightarrow \infty$, to determine P_{fa}



- For Simulation
  - Count the number of missed detections for "on-time" position
  - Count the number of false alarms for "off-time" position

Performance of 1-D System in terms of P_{md} and P_{fa}

- P_{fa}= Probability of False Alarm
- P_{md}= Probability of Missed Detection
- K=200 users,
- $\tilde{d} = 0.7d \rightarrow \sigma_x^2 = 0.09$ (data reliability)
- Processing Gain (N=100)
- Conventional Acquisition fails
- Partial cancellation performance result is acceptable



Acquisition- Summary

- High performance MUD designs are capacity limited by conventional acquisition techniques
- Sharing of “soft” information between acquisition unit and receiver are essential to maximise capacity (Need to use turbo principle)
- Based on known receiver performance (BER), our proposed technique significantly improves performance
- An acquisition technique like this is essential for MUD operation
- Similiar techniques can be used for frame acquisition and tracking tasks

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MAP Algorithm (1) -Introduction

- The MAP algorithm provides soft outputs for coded and information bits
- For a convolutional code the decoding structure is a trellis diagram
- The MAP algorithm can also be used for block codes [7] (not covered here)
- The APP Algorithm calculates all transition probabilities and determines the liklihood that a particular bit was sent
- The maximisation of the APP decoder is the MAP and this is the optimal decoder.
- The MAP decoder alone has the same performance as a Viterbi Decoder(VD) [8].
- The MAP decoder within a turbo receiver/decoder has substantial better performance than a VD.

MAP Algorithm (2) -The Steps

The algorithm steps in Summary are:-

1. Compute metrics for each bit transmitted
2. Compute transition probabilities
3. Compute forward state probabilities
4. Compute reverse state probabilities
5. Determine the joint transition probabilies
6. Determine the probability of a information data bit by summing over joint transition probabilies
7. Determine the probability of a coded data bit by summing over joint transition probabilies

MAP Algorithm (3) -Definitions

- Forward State Probability: $\alpha_j(m) = \Pr\{S_j = m, \mathbf{y}_1^j\}$
- Reverse State Probability: $\beta_j(m) = \Pr\{\mathbf{y}_{j+1}^{T_u} | S_j = m\}$
- Transition Probability: $\gamma_j(m', m) = \Pr\{S_j = m, \mathbf{y}_j | S_{j-1} = m'\}$
- Joint Trans. prob.: $p\{S_{j-1} = m', S_j = m, \mathbf{y}\} \approx p\{S_{j-1} = m', S_j = m | \mathbf{y}\}$

Joint Transition probability:

$$\begin{aligned} \Pr\{S_{j-1} = m', S_j = m | \mathbf{y}\} &= \Pr\{S_{j-1} = m', b_j = b | \mathbf{y}\} \\ &= \Pr\{S_{j-1} = m', \mathbf{d}_j = \mathbf{d} | \mathbf{y}\} \end{aligned}$$

MAP Algorithm (4) -The Equations

1. Metrics for coded bit probabilities: $\Pr\{y_{j,n'}|d_{j,n'}\} = \frac{1}{\sigma\sqrt{2\pi}} \exp\left(-\frac{(y_{j,n'}-d_{j,n'})^2}{2\sigma^2}\right)$

2. Transition Probability: $\gamma_j(m',m) = \Pr\{S_j = m|S_{j-1} = m'\} \prod_{n'=1}^{n} \Pr\{y_{j,n'}|d_{j,n'}\}$.

3. Forward State Probability: $\alpha_j(m) = \sum_{m'=0}^{M-1} \alpha_{j-1}(m')\gamma_j(m',m)$

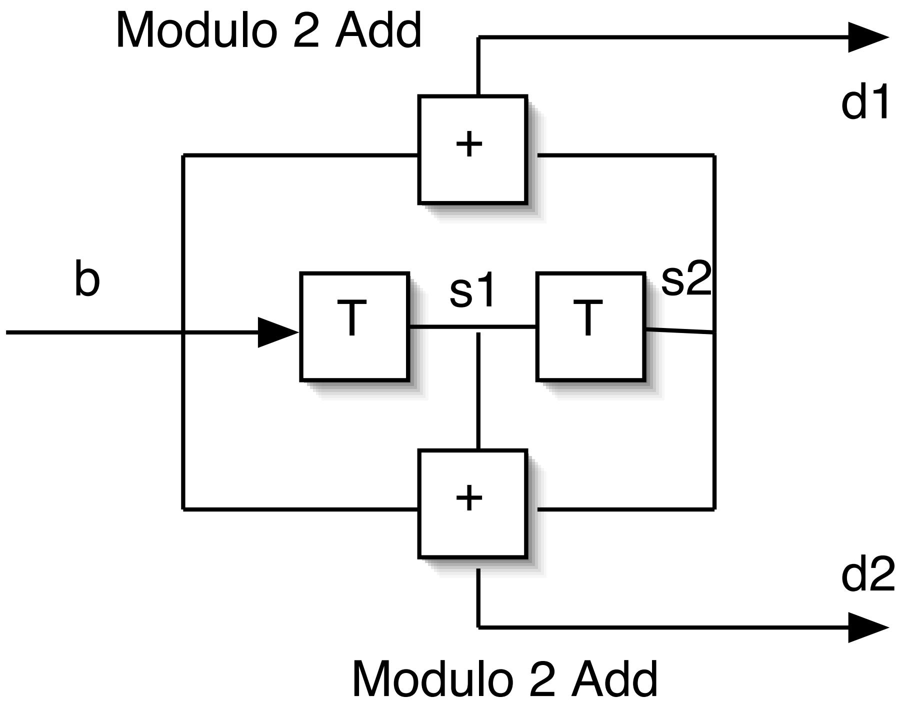
4. Reverse State Probability: $\beta_j(m) = \sum_{m'=0}^{M-1} \beta_{j+1}(m')\gamma_{j+1}(m,m')$

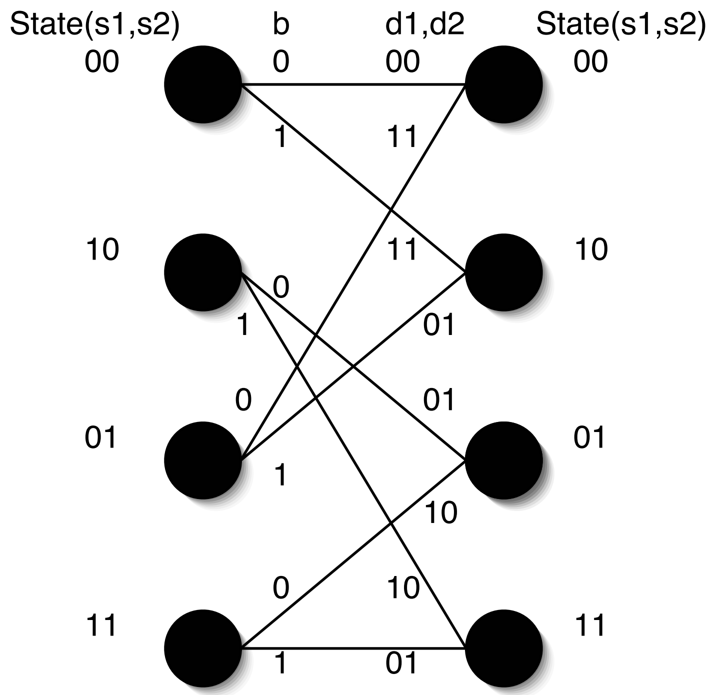
5. Joint Trans. prob.: $p\{S_{j-1} = m', S_j = m, \mathbf{y}\} = \alpha_{j-1}(m')\gamma_j(m',m)\beta_j(m)$

6. Prob. coded bit trans.: $\Pr\{d_{j,n'} = d|\mathbf{y}\} = \sum_{m'} \sum_{\substack{\mathbf{d} \\ (d_{j,n'}=d)}} \Pr\{S_{j-1} = m', \mathbf{d}_j = \mathbf{d}|\mathbf{y}\}$

7. Probability of information bit transition: $\Pr\{b_j|\mathbf{y}\} = \sum_{m'} \Pr\{S_{j-1} = m', b_j = b|\mathbf{y}\}$.

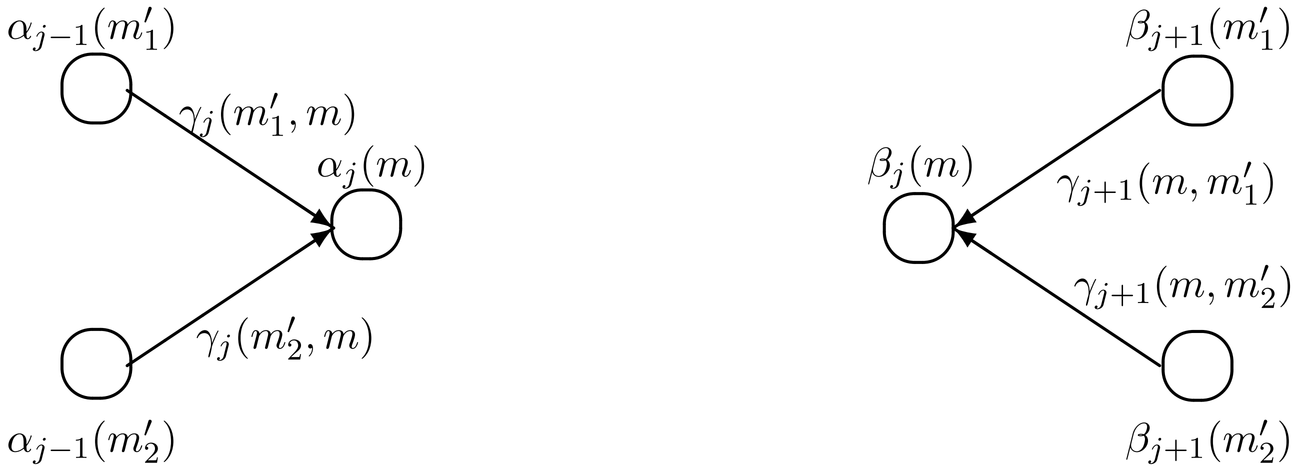
MAP Algorithm-Encoder/Trellis Example





- Convolutional Encoder is G[5,7], Rate 1/2, Memory 2 (4 State)
- Single Trellis transition shows initial state, possible transitions,next state
- Values for information bit *(b), coded bits (d1,d2) are shown*

MAP Algorithm Example (1)



- The Forward and Reverse state probability calculation
- After each transition the values need to be normalised to stop overflow
- Alternative methods such as LogMAP exist for implementation in Hardware [9]

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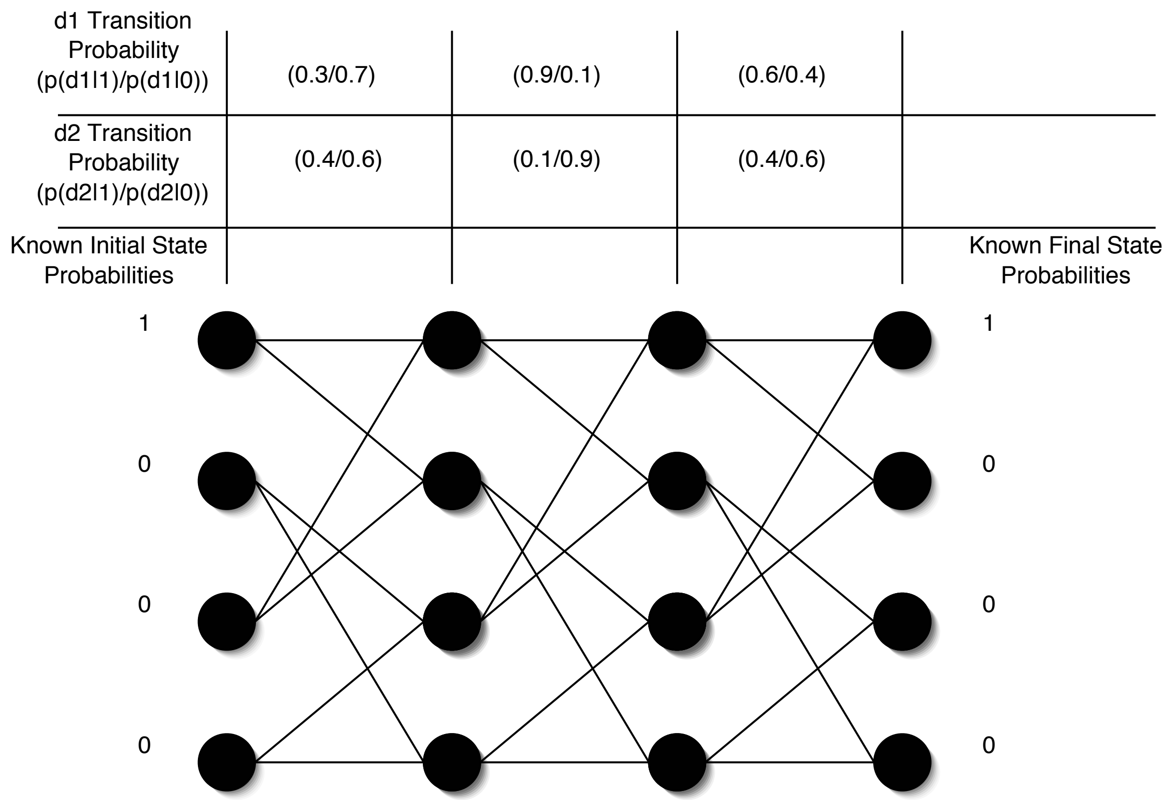
MAP Algorithm Example (2) - An Exercise

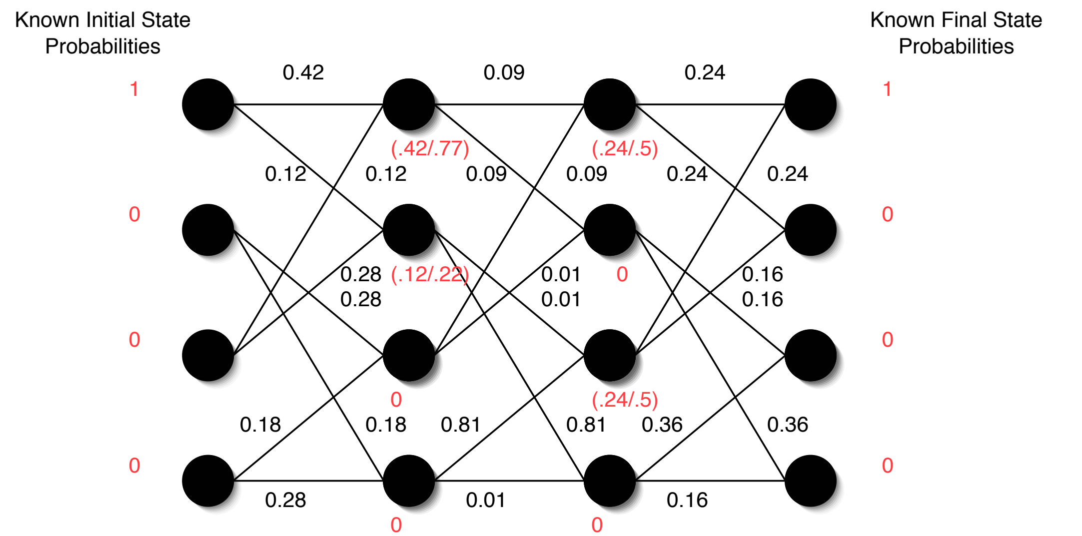
Here is a four state trellis based on a G[5,7] conv. encoder (shown earlier).

Determine the APP Decoder outputs based on the information given.



MAP Algorithm Example (3) - Solution to Exercise

- Normalisation of probabilities at every step
- Probabilities for coded and information bits is shown



00->0.42	00->0.09	00->0.24
01->0.28	01->0.01	01->0.16
10->0.18	10->0.81	10->0.36
11->0.12	11->0.09	11->0.24

State Probabilities (Red)
Transition Probabilities (Black)

P(d1=1ly) = 0.0264	P(d1=1ly) = 0	P(d1=1ly) = 0.12
P(d1=0ly) = 0.3234	P(d1=0ly) = 0.0357	P(d1=0ly) = 0.12
P(d2=1ly) = 0.0264	P(d2=1ly) = 0.0011	P(d2=1ly) = 0.12
P(d2=0ly) = 0.3234	P(d2=0ly) = 0.0346	P(d2=0ly) = 0.12
P(b=1ly) = 0.0.0264	P(b=1ly) = 0	P(b=1ly) = 0
P(b=0ly) = 0.3234	P(b=0ly) = 0.03575	P(b=0ly) = 0.24

Hard Decision (info bits)	0	0	0
Hard Decision (coded bits (d1/d2))	(0/0)	(0/0)	(?/?) (?= either 1 or 0)

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References

[1] P. D. Alexander, A. J. Grant, and M. C. Reed, “Performance analysis of an iterative decoder for code-division multiple-access,” *European Trans. on Telecom.*, vol. 9, pp. 419–426, Sep./Oct. 1998.

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[7] J. Hagenauer, “The turbo principle:Tutorial introduction and state of the art,” in *Int. Symp. on Turbo Codes and Applications*, (Brest, France), pp. 1–11, Sept. 1997.

[8] L. R. Bahl, J. Cocke, F. Jelinek, and J. Raviv, “Optimal decoding of linear codes for minimising symbol error rate,” *IEEE Trans. Inform. Theory*, vol. 20, pp. 284–287, 1974.

[9] P. Robertson, E. Villebrun, and P. Hoeher, “A comparison of optimal and sub-optimal MAP decoding algorithms operating in the log domain,” in *IEEE Int. Conf. on Communications*, ((Seattle, U.S.A.)), pp. 1009–1013, Jun. 1995.

Overview of Wireless Communications
MIMO

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March 2004

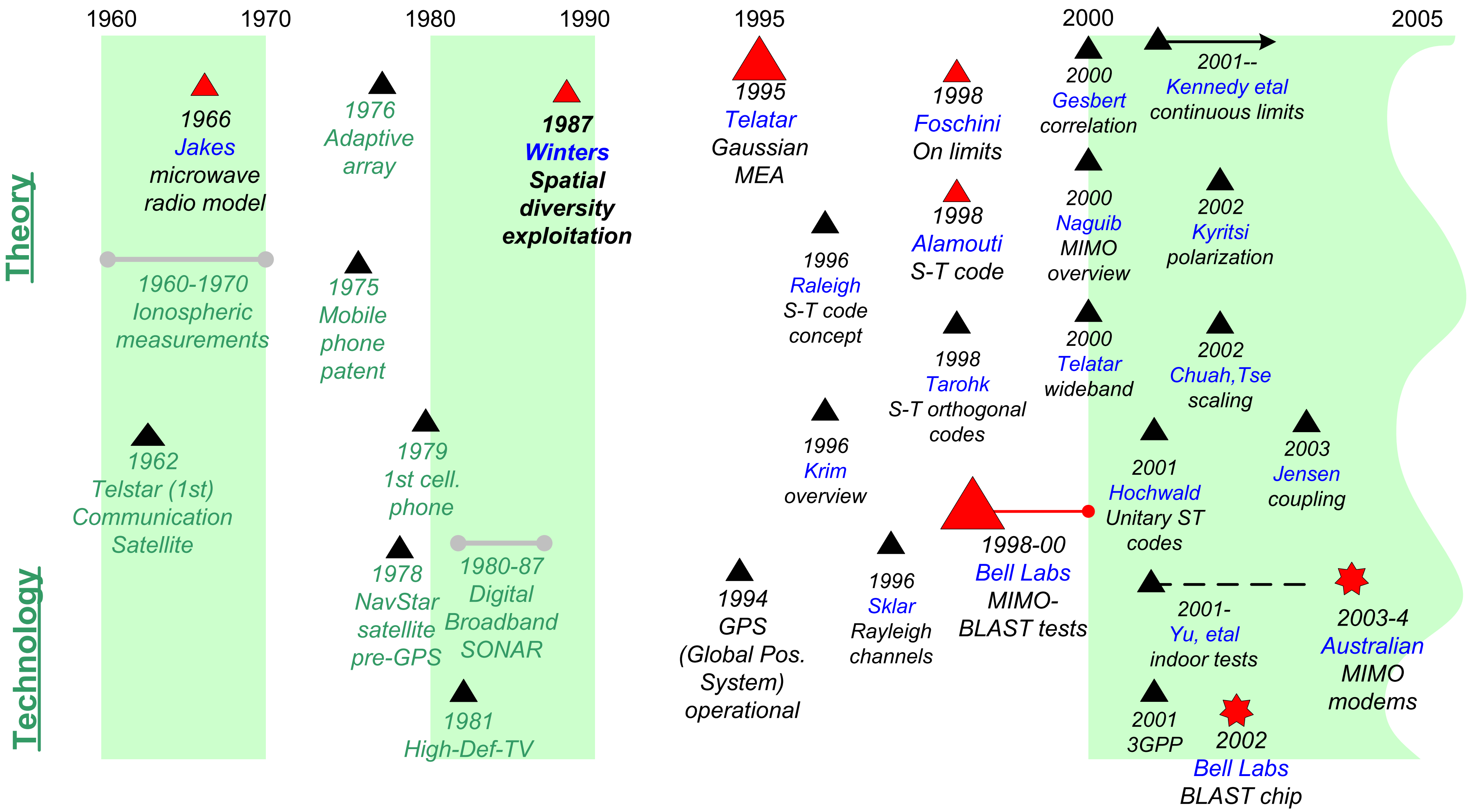
MIMO Overview

- Day 1
  - Introduction - Motivation and Background details
  - History
  - Beamforming
  - Multipath environments
  - Multiple-input, multiple-output concept
  - Information Theory for matrix channels
- Day 2
  - Extensions
  - Non-ideal environments
  - Channel models
  - Coding
  - Implementation

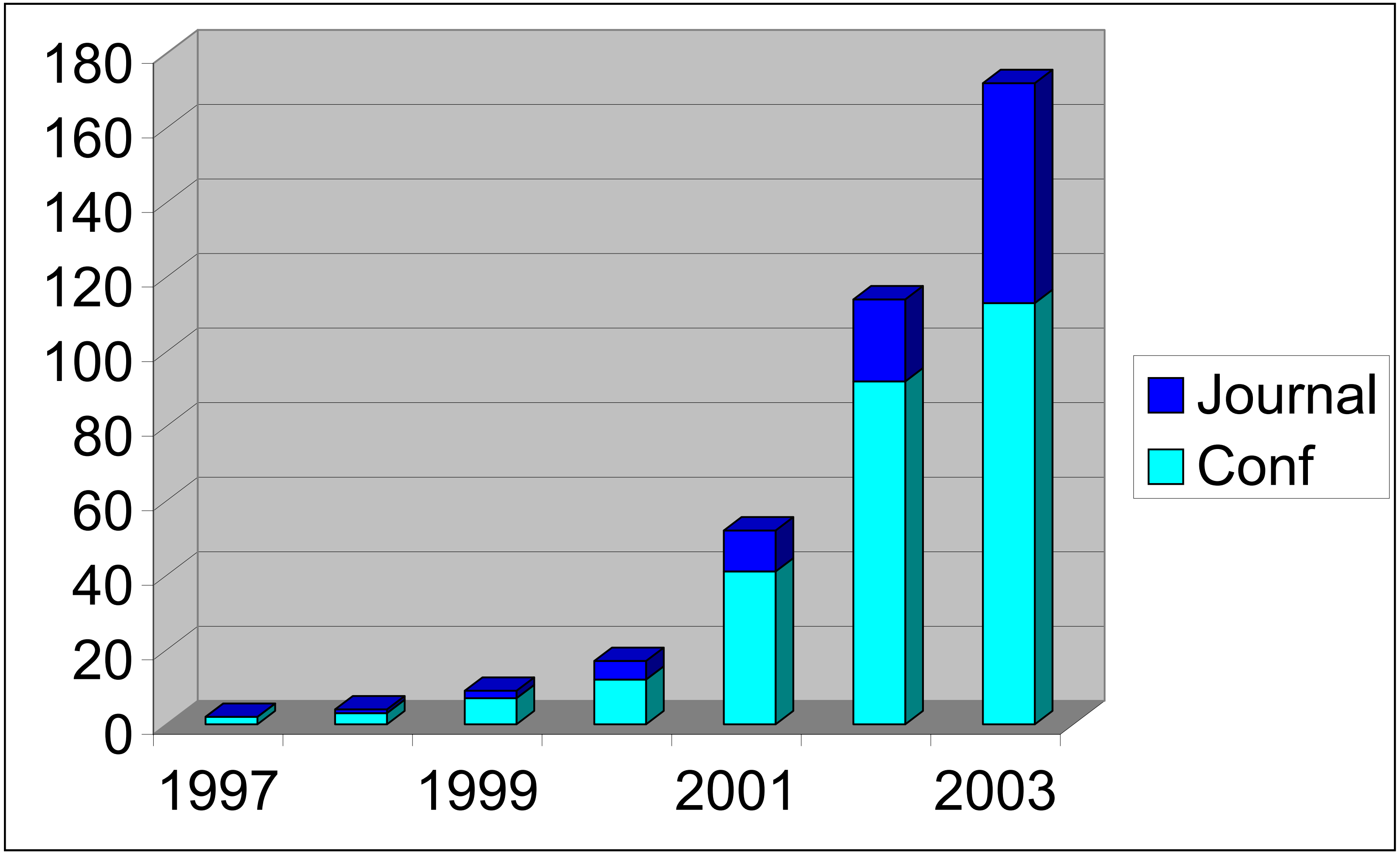
MIMO Day 1

- Introduction - Motivation and Background details
- Multiple-input, multiple-output concept
- By the end of today, you should
  - Understand beamforming terminology, eg. “steering vectors”
  - Understand what multipath is, and how it is mitigated
  - Know what MIMO is
  - Be able to derive basic MIMO information theoretic results

World-wide interest in MIMO



World-wide interest in MIMO (cont)

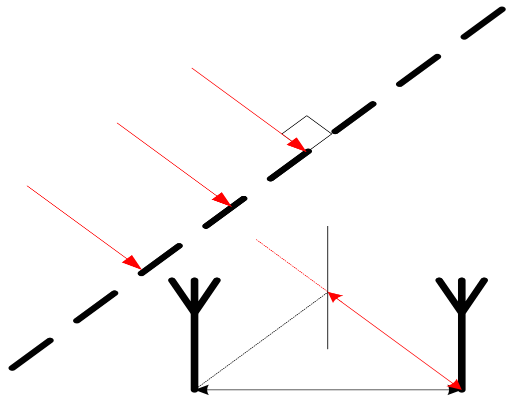


MIMO publications since 1996[1]

[1]Search on IEEE-Xplore, using selected keywords, in Comms conferences & journals.

Historical Background: Beamforming

- Beamforming arose from RADAR technology in 1960's
  - used in sonar, acoustics, EM broad-band and narrow-band
  - "Beam" used to focus transmission/reception of signals according to location.

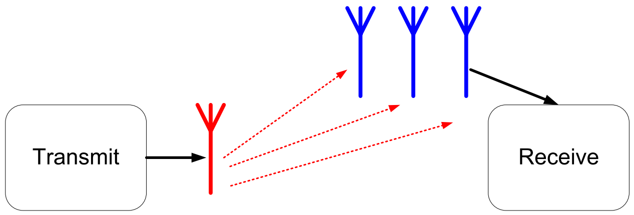


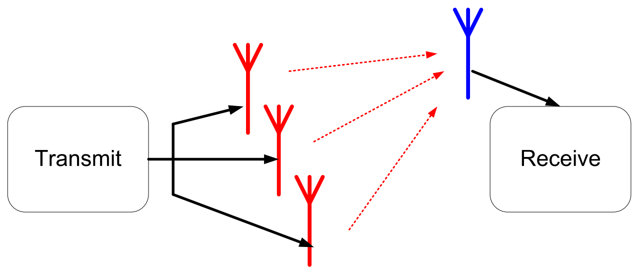
[KV96] H. Krim and M. Viberg. Two decades of array signal processing. *IEEE Signal Processing Mag.*, pages 67–94, July 1996.

[VVB88] B. D. Van Veen and K. M. Buckley. Beamforming: a versatile approach to spatial filtering. *IEEE ASSP Mag.*, 5(2):4 – 24, April 1988.

[PP97] A. J. Paulraj and C. B. Papadias. Space-time processing for wireless communications, improving capacity, coverage, and quality in wireless networks by exploiting the spatial dimension. *IEEE Signal Processing Mag.*, pages 49–83, November 1997.

Beamforming, signal benefits





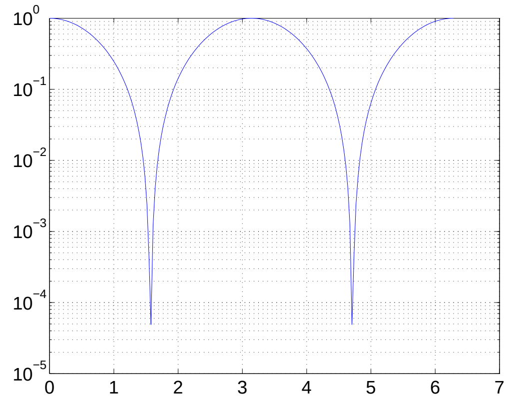
- Single-input, multiple-output
  - Each rx-antenna has independent noise, equal variance (i.i.d.)
  - Each rx-antenna r receives signal power $P_r \approx P_t \frac{1}{D^2} g_r g_t \cdot h_{1\to r}$
  - ⋆ Total signal power $P \propto R \cdot P_t$
  - Noise power only σ^2 (independent)
  - SNR $\propto RP_t/\sigma^2$
  - Rx can "listen" to one point in space

- Multiple-input, single-output
  - Each tx-antenna t sends signal power P_t/t
  - ⋆ Total signal power $P \propto TP_t$ *if coherent*
  - Noise power σ^2
  - SNR $\leq TP_t/\sigma^2$
  - Tx can focus on one point in space

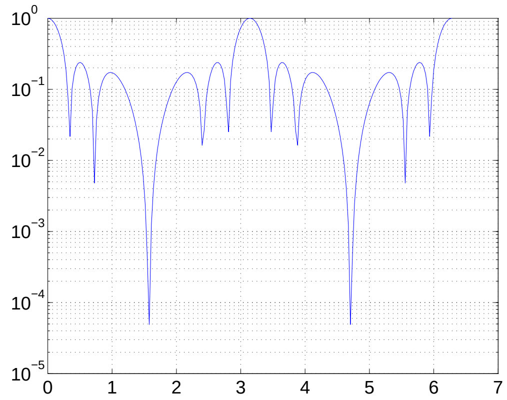
Potential diversity of $T \cdot R$

Beamforming;

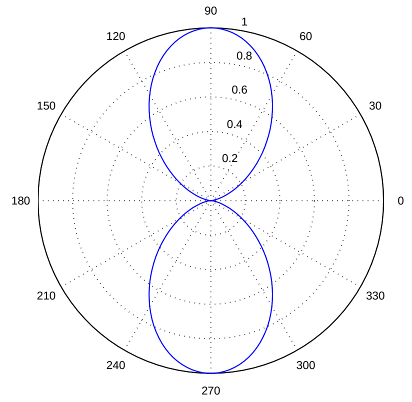
- Linear array is simplest beamforming system: concepts carry across easily to other array geometries.
- Element spacing gives phase offsets across array
- Increasing number of elements gives sharper beam
- “Natural” beam shape shown



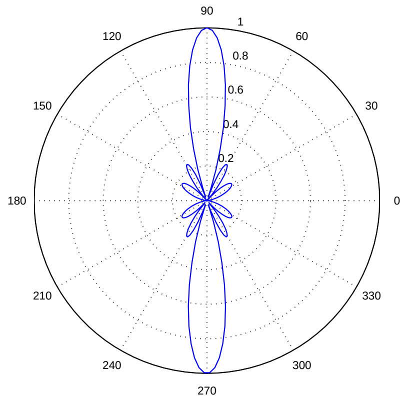
Array magnitude response, vs angle, 2elem array



Array magnitude response, vs angle, 6elem array



Array magnitude response, vs angle, 2elem array



Array magnitude response, vs angle, 6elem array

Beamforming; vector channel

- Each element of beamformer has (desired) signal plus noise. $\mathbf{r}_m = \mathbf{s}_m + \mathbf{z}_m$
- Output of beamformer is weighted sum $y = \sum_{m=1}^{M} \mathbf{w}_m \left(\mathbf{s}_m + \mathbf{z}_m\right)$
- Vector notation: $y = \mathbf{w}\left(\mathbf{s} + \mathbf{z}\right)$
- What is SNR?

$$E\left\{\mathrm{P}_{\mathrm{Rec}}\right\} = E\left\{y\bar{y}\right\} == E\left\{\mathbf{w}\mathbf{s}\mathbf{s}^{\dagger}\mathbf{w}^{\dagger} + \mathbf{w}\mathbf{z}\mathbf{z}^{\dagger}\mathbf{w}^{\dagger}\right\}$$

$$\mathrm{SNR} = \frac{\mathbf{w}\mathbf{S}\mathbf{w}^{\dagger}}{\mathbf{w}\mathbf{R}\mathbf{w}^{\dagger}}$$

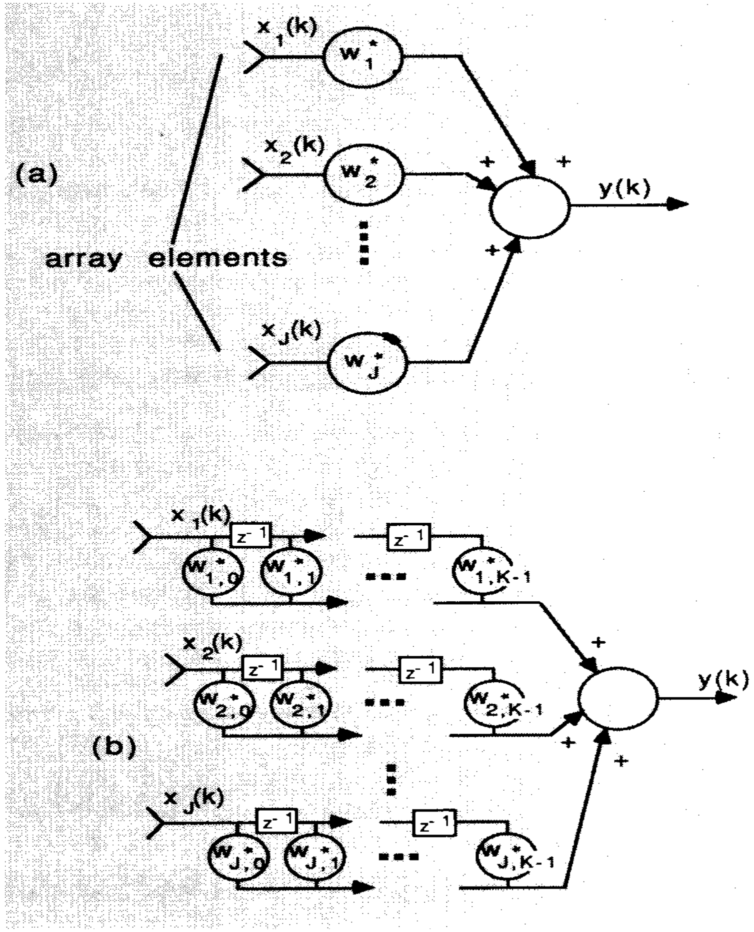
- Optimal MMSE beamformer chooses $\mathbf{w}$ to *minimise* SNR.

By selecting the weight vector $\mathbf{w}$ we can alter the “shape” of the beam, to a particular desired response. This response may be simple eg. “steering the beam” to simulate physical rotation of the array, or more complex. In each case, $\mathbf{w}$ is referred to as a “steering vector.”

Beamforming; extensions

- Can extend from single-freq to broadband
  - must consider beamformer as multiple FIR filters.

- Choice of “steering vectors”
  - chase desired signal
  - steer nulls at noise
  - adaptive

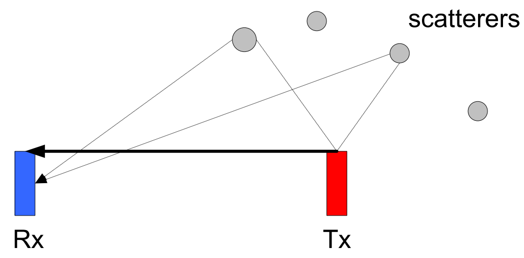


Single-freq and Broadband beamforming [VVB88]

Multipath

- Signal $s(t)$ arrives at multiple times (echoes)[Pro89] with varying amplitudes.

$$r(t) = \sum_{l=1}^{L} \alpha_l(t) s\left(t - \tau_n(t)\right)$$

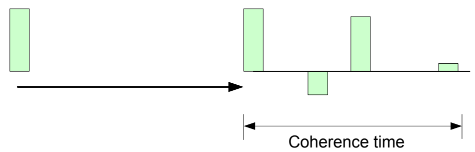


General multipath

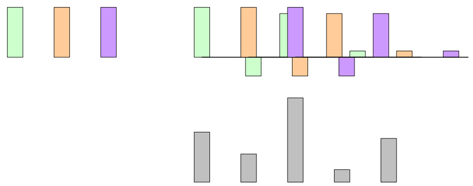
- Simpler: consider unmodulated narrow-band carrier.

$$r(t) = \sum_{l=1}^{L} \alpha_l(t) e^{-\jmath 2\pi f \tau_l(t)}$$

$$r(t) = \int_L \alpha(t,l) e^{-\jmath 2\pi f \tau(t,l)}\, dl$$



Single narrow-band signal



Inter-symbol interference

[Pro89] J. G. Proakis. *Digital Communications*. Computer Science Series. McGraw-Hill, New York, USA, 2nd edition, 1989.

Multipath; models

- Frequency selective
  - Signals arrive at different times
  - tap-delay line

$$y_k = \sum_{l=1}^{L} \alpha_l x_{k-\Delta} + z_k$$

- Frequency flat
  - Signals arrive together
  - single scalar gain

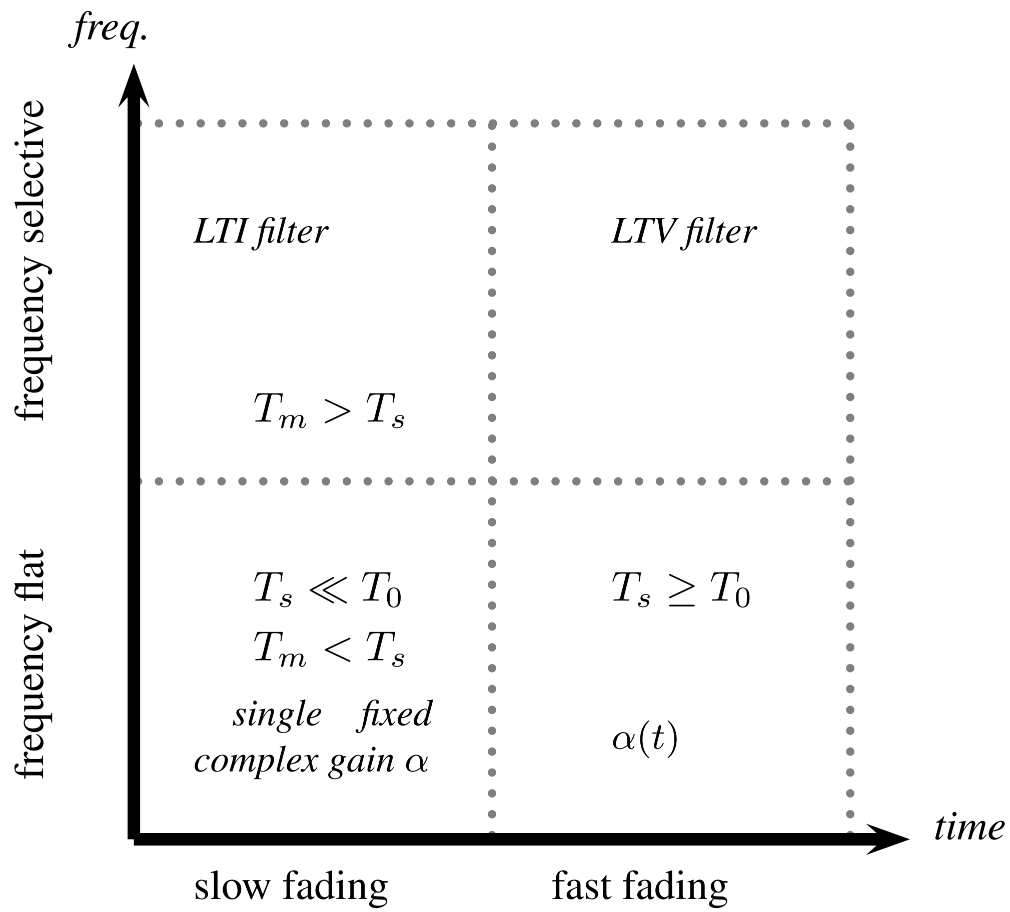
$$y_k = \alpha_k x_{k-\Delta} + z_k$$

Models choose coefficients α

- Fast/Slow fading – speed at which α change.
- Geometric: *multipath caused by "little dots"*
- Stochastic: *random variable, according to a distribution*
- Measurement-based: *Some given data, matched to channel*

We shall consider *slow-fading, frequency-flat* models

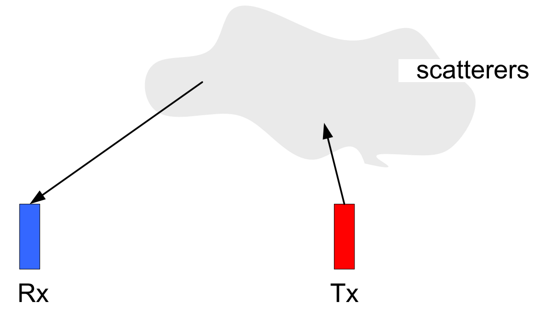
Multipath; SISO model map



With permission from [Han03]

Stochastic Multipath; Rayleigh model

Simplest stochastic model.
Arises from ionospheric measurements

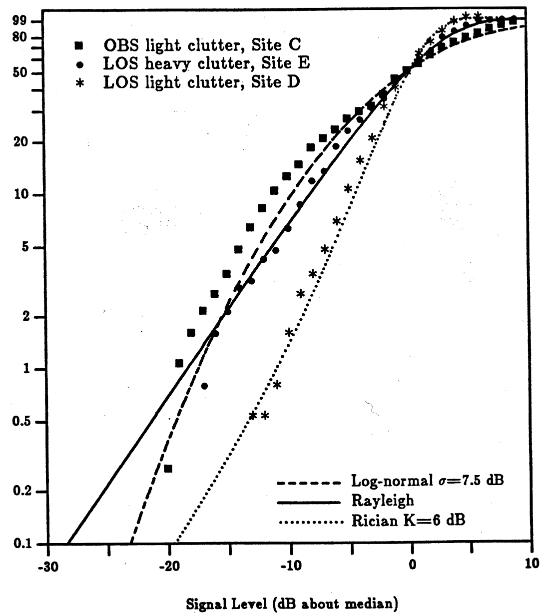


- Receive signal

$$r = \alpha = g e^{\jmath\theta} = \sum (a_i + \jmath b_i) \approx a + \jmath b$$

- a and b are zero-mean, Gaussian r.v.'s (law-large-numbers)

- $|\alpha|$ is *Rayleigh distributed*



From [Rap02]

Other options include *Ricean* (non-zero mean), *Log-normal* and many others. Each model is tailored for the environment of interest.

Multipath; mitigation

- Most works (pre-1995) focus on removing effects of multipath[Skl97]
- Multipath mitigation generates many channel models
- Fading is *time-varying multipath*

- *Estimate multipath channel*
- *Consider initial signal "good" everything else unwanted noise*
- *Use temporal signal char's to remove ISI*
- *Use spatial signal char's to remove additional*
- *Iterate if desired*

[Skl97] B. Sklar. Rayleigh fading channels in mobile digital communication systems part II: Mitigation. *IEEE Commun. Mag.*, pages 105–112, July 1997.

Multipath mitigation; diversity

- Diversity: if some part of the channel is bad (sometimes) then use ensemble

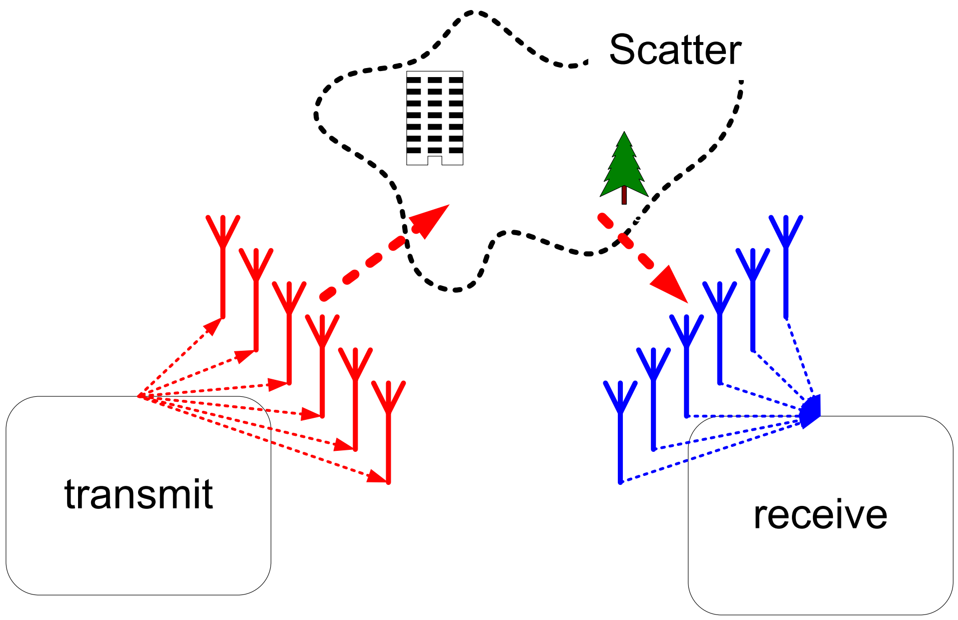
- Time-diversity:
  - Channel may fade (drop-out)
  - spread bits over time (recall previous lectures!)

- Frequency-diversity:
  - doppler, ISI, frequency-selective fades
  - spread frequency of signal (eg. CDMA, UWB)

- Spatial-diversity:
  - multipath comes from different angles
  - steer a beam toward the good angles
  - average over all angles to “stabilize” channel.

MIMO Concept

All that multipath contained signals... why not use it?

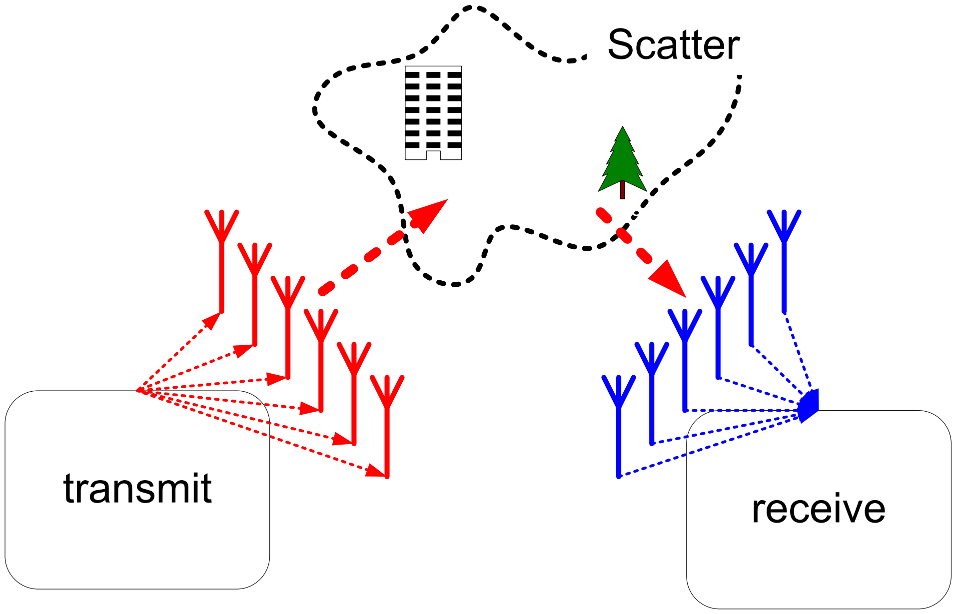


We will shortly delve into matrix channels....

"Unfortunately, no-one can be told what the channel is, you have to see it for yourself."

[WSG92] J. H. Winters, J. Salz, and R. D. Gitlin. The capacity of wireless communication systems can be substantially increased by the use of antenna diversity. In *1st International Conference on Universal Personal Communications, (ICUPC '92) Proceedings.*, pages 02.01/1 – 02.01/5, 29 September –1 October 1992.

MIMO Big claims



Jack H. Winters

- Capacity of channel increases linearly with number of elements.
- Coding is possible, and not *too* complex
  - Some early results showed huge improvements without coding

[WG94] J. H. Winters and M. J. Gans. The range increase of adaptive versus phased arrays in mobile radio systems. In *Twenty-Eighth Asilomar Conference on Signals, Systems and Computers*, volume 1, pages 109 – 115, 31 October –2 November 1994.

[Win94] J. H. Winters. The diversity gain of transmit diversity in wireless systems with rayleigh fading. In *IEEE Intl. Conf. on Commun., ICC'94 and SUPERCOMM'94 Serving Humanity Through Communications.*, volume 2, pages 1121 – 1125, May 1–5 1994.

Parallel, Additive White Gaussian, Channels[Gal68]

Robert G. Gallager

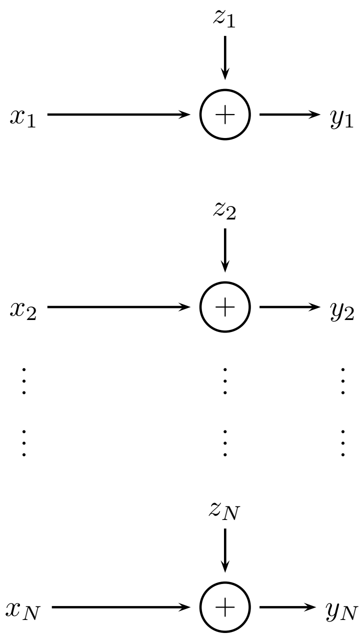
- Consider N channels, which are independent, discrete, parallel, AWGN.

  We have N inputs, power limited, so that $\sum_{n=1}^{N} |x_n|^2 \leq P$. Noise in channel n is *white, Gaussian* with variance (power) $E\left\{|z_n|^2\right\} = \sigma_n^2$

- Recall:

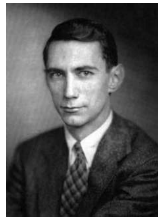
  Capacity of single channel ($N = 1$) with transmit power P and σ^2 noise

$$\mathcal{I}\left(X_1; Y_1\right) \leq C \leq \log\left(1 + \frac{P}{\sigma^2}\right)$$



[Gal68] R. Gallager. *Information Theory and Reliable Communication*. John Wiley & Sons, New York, USA, 1968.

Parallel, Additive White Gaussian, Channels

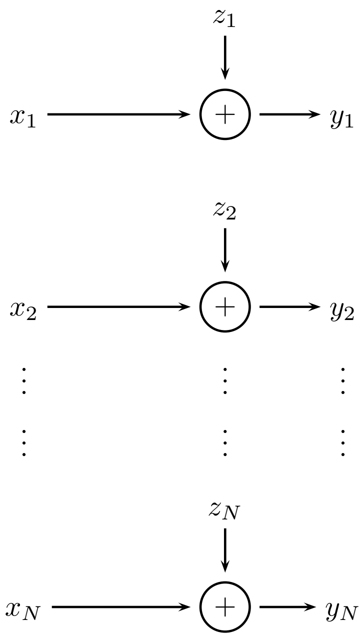


Claude E. Shannon

- We can consider the entire ensemble of inputs $\mathbf{X}^N = \{X_1, \ldots, X_N\}$ and entire ensemble of outputs $\mathbf{Y}^N = \{Y_1, \ldots, Y_N\}$

$$\mathcal{I}\left(\mathbf{X}^N; \mathbf{Y}^N\right) \leq \sum_{n=1}^{N} \mathcal{I}\left(X_n; Y_n\right) \leq \sum_{n=1}^{N} \log\left(1 + \frac{P_n}{\sigma_n^2}\right)$$

$$\sum_{n=1}^{N} P_n \leq P$$



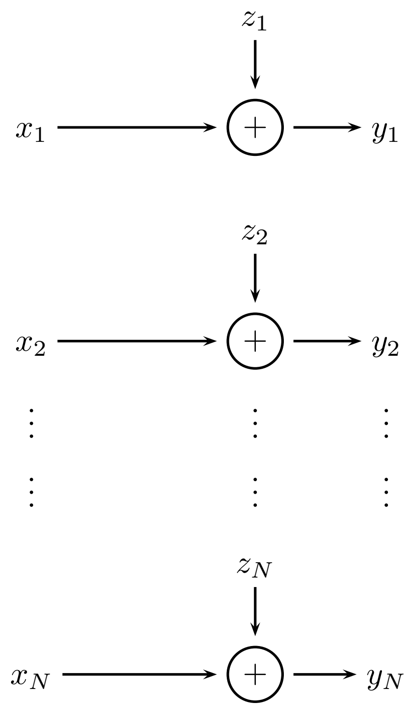
- Equality iff x_n & z_n are independent Gaussian[Sha48]. Gallager uses 1/2, this corresponds to real r.v.'s
- ***Question:*** Can we simplify this?

[Sha48] C. E. Shannon. A mathematical theory of communication. *Bell System Tech. J.*, 27:379–423, 623–656, July 1948.

Parallel, Additive White Gaussian, Channels

- What about equal power noise?

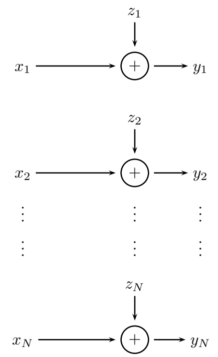
$$\sigma_n^2 = \sigma^2 \quad \forall n$$



Does equal power noise mean we get identical noise samples?

Parallel, ... Waterfilling

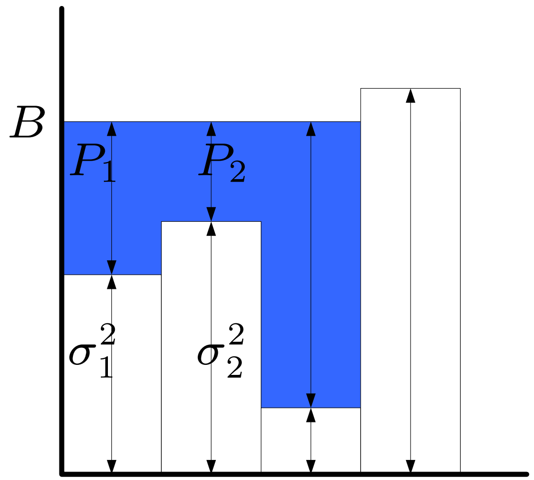
- Assume transmitter knows channel. Assume noise has non-equal powers.
  - Assign more power to "better" channels.
  - Must satisfy power constraint
- Solution: Waterfill. Choose x_n as independent Gaussians, with variance (power) given by P_n such that:



$$P_n = \begin{cases} B - \sigma_n^2; & \sigma_n^2 < B \\ 0; & \text{otherwise} \end{cases}$$

$$C = \sum_{n=1}^{N} \log_2 \left(1 + \frac{P_n}{\sigma_n^2} \right) \quad \text{bit/s/Hz}$$

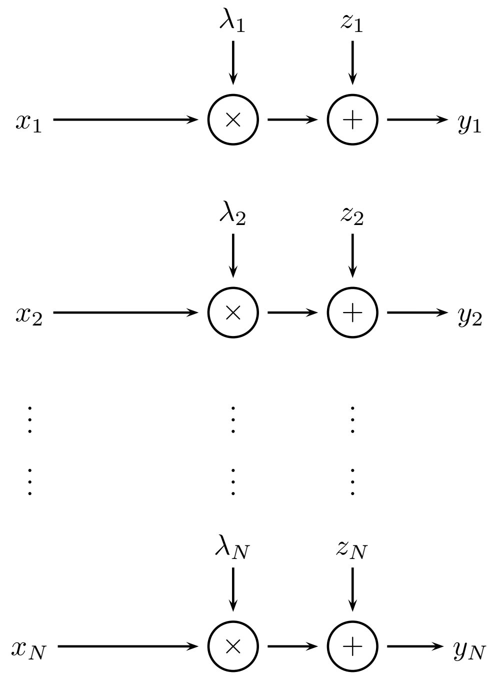
$$= \sum_{n:\sigma_n^2 \le B} \log_2 \left(\frac{B}{\sigma_n^2} \right)$$



Parallel AWGN channels with gains

- Assume transmitter knows channel.
- Assume noise has non-equal powers.
- Assume each channel also has a gain λ_n
  - Assign more power to "better" channels.
  - Must satisfy power constraint

$$C \leq \sum_{n=1}^{N} \log\left(1 + P_n \frac{\lambda_n^2}{\sigma_n^2}\right) \quad (1)$$



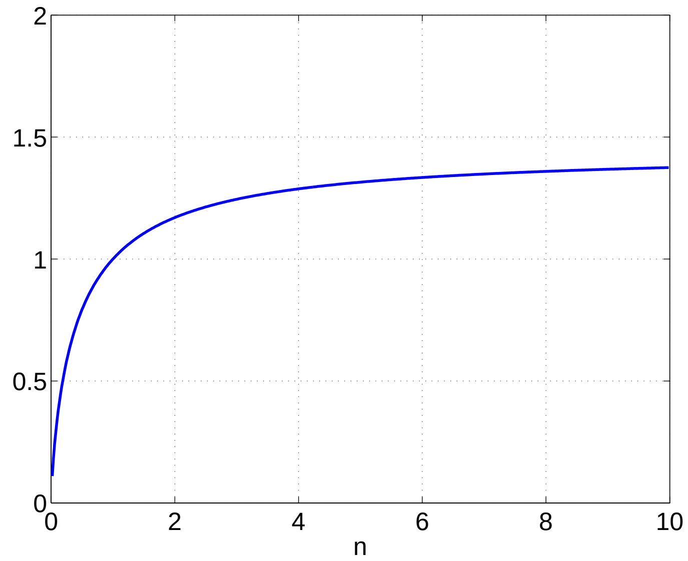
- Solution?

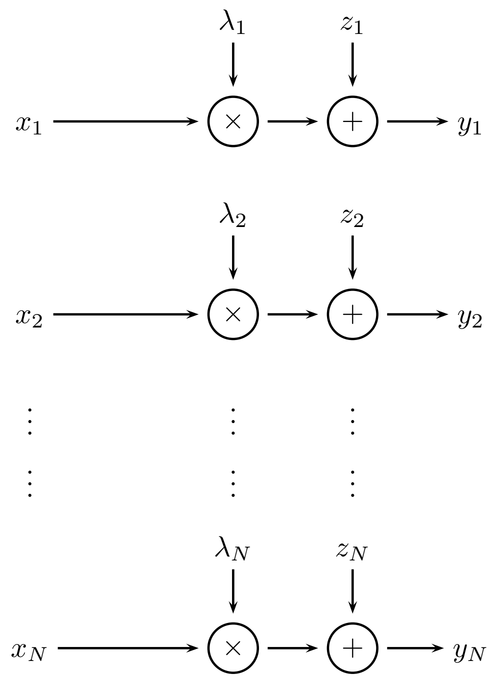
Is this such an amazing result? It was known in 1968.

- Assume transmitter knows channel. $\sigma_n^2 = 1$. $\lambda_n = 1$

$$C \leq \sum_{n=1}^{N} \log\left(1 + P_n \frac{\lambda_n}{\sigma_n^2}\right) \quad P_n = \frac{P}{N}, \sigma_n^2 = 1, \lambda_n = 1$$

$$= N \log\left(1 + \frac{P}{N}\right) \leq P \log_2 e \quad \text{bit/s/Hz}$$





The world pre-MIMO

- Single channel.
  - Capacity proportional to log of power.

$$\boxed{C = \log(1 + P) \qquad (2)}$$

  - lower SNR reduces capacity – equivalent to less power.
- Parallel channels.
  - Waterfilling gives capacity
  - larger gains improve capacity, noise reduces capacity
  - Best capacity is equal gain, equal noise, gives

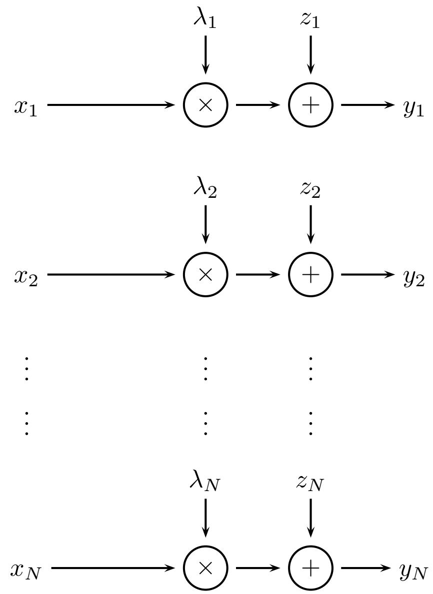
$$\boxed{C \le N \log\left(1 + \frac{P}{N}\right) \le P \log e \qquad (3)}$$

MIMO preliminaries

- How can we arrange the parallel channels neatly?
  - collect all inputs and outputs into vectors.

$$\begin{bmatrix} y_1 \\ y_2 \\ \vdots \\ y_N \end{bmatrix} = \begin{bmatrix} \lambda_1 & & & \\ & \lambda_2 & & \\ & & \ddots & \\ & & & \lambda_N \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_N \end{bmatrix} + \begin{bmatrix} z_1 \\ z_2 \\ \vdots \\ z_N \end{bmatrix}$$

$$\mathbf{y} = \Lambda \mathbf{x} + \mathbf{z} \tag{4}$$



  - Read off each row to get parallel channels
  - Power limit becomes

$$E\left\{ \mathrm{Tr}\left(\mathbf{x}\mathbf{x}^{\dagger} \right) \right\} \leq P$$

  - Matrix $\mathbf{Q} = E\left\{\mathbf{x}\mathbf{x}^{\dagger}\right\}$ called “covariance” of input signal.

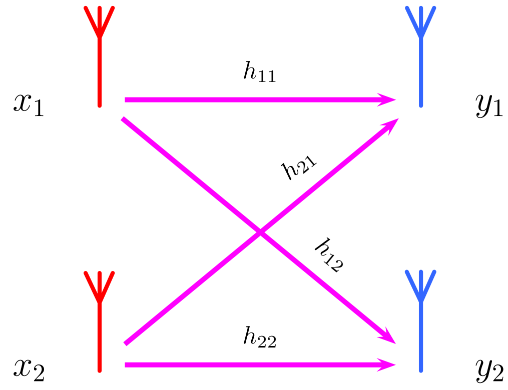
MIMO prelim. toward real systems

- Each receiver (blue) detects signals from *all* transmitters (red) plus iid noise

$$y_r = \sum_{t=1}^{T} h_{tr} x_t + z_r$$

$$\mathbf{y} = \mathbf{H}\mathbf{x} + \mathbf{z} \quad \mathbf{H} = \begin{bmatrix} h_{11} & h_{21} & \cdots & h_{t1} \\ h_{12} & h_{22} & \cdots & h_{t2} \\ \vdots & \vdots & \ddots & \vdots \\ h_{1r} & h_{2r} & \cdots & h_{tr} \end{bmatrix} \tag{5}$$

T transmit, R receive.

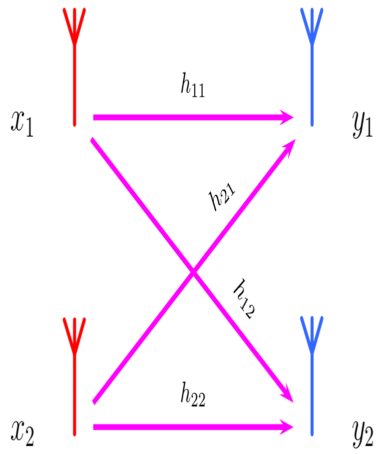


- Note similarity to previous "vector" channel.
  - h_{ij} gives *complex* gain from transmit i to receiver j. Be careful of notation!
  - We will assume receiver has full channel knowledge. Is this reasonable?

MIMO prelim. Full tx knowledge

- Transmitter “diagonalises” channel $\mathbf{y} = \mathbf{H}\mathbf{x} + \mathbf{z}$
  - transmit signals in certain way, to ensure no cross-talk.
  - Basic rule: capacity determined by SNR, need white noise. Invariant transforms....
- Unitary transforms are invariant for Gaussian processes
  - SVD of channel, $\mathbf{H} = \mathbf{U}\mathbf{\Lambda}\mathbf{V}^{\dagger}$. $\mathbf{\Lambda}$ contains *singular* values, not eigenvalues.
  - Receiver applies “filter” $\hat{\mathbf{y}} = \mathbf{U}^{\dagger}\mathbf{y}$
  - Transmitter modifies distribution $\hat{\mathbf{x}} = \mathbf{V}\mathbf{x}$
  - “New channel” $\hat{\mathbf{y}} = \mathbf{\Lambda}\hat{\mathbf{x}} + \hat{\mathbf{z}}$

$$C = \sum_{m=1:\frac{\sigma_m^2}{\lambda_m^2} \leq B}^{M=\min(R,nt)} \log\left(B\frac{\lambda_m^2}{\sigma_m^2}\right) \quad (6)$$



MIMO prelim. Full tx knowledge [Tel99]

- $\mathbf{y} = \mathbf{H}\mathbf{x} + \mathbf{z}$
  - Assume unitary noise $\sigma^2 = 1$. (We adjust P arbitrarily)
  - Covariance of received signal $E\left\{\mathbf{y}\mathbf{y}^\dagger\right\} = \mathbf{HQH}^\dagger + \mathbf{I_r}$
  - mutual info $\mathcal{I}\left(\mathbf{x};\mathbf{y}\right) = \sum_n \log\left(1 + P_n\lambda_n\right)$
- BUT.
  - $\sum_j \log(a_j) = \log\left(\prod_j a_j\right)$
  - $\det(X) = \prod_j \lambda_j$ where λ is eigenvalues of X.

$$\mathcal{I}\left(\mathbf{x};\mathbf{y}\right) = \log\det\left(I + \mathbf{HQH}^\dagger\right)$$

  - Transmitter gets to choose $\mathbf{Q}$.

$$C = \sup_{\mathbf{x}:\mathrm{Tr}(\mathbf{Q})\leq P} \mathcal{I}\left(\mathbf{x};\mathbf{y}\right) = \sup_{\mathbf{Q}:\mathrm{Tr}(\mathbf{Q})\leq P} \log\det\left(I + \mathbf{HQH}^\dagger\right) \tag{7}$$

I. Emre Telatar

Jerry G. Foschini

Thomas Marzetta

[Tel99] I. E. Telatar. Capacity of multi-antenna gaussian channels. *Euro. Trans. Telecomm.*, 10(6):585–595, November 1999.

MIMO for real now...

- What if transmitter doesn't know channel? (Is this important?)
  - Receiver still knows $\mathbf{H}$, so we can calculate mutual information
  - $\mathcal{I}(\mathbf{x}; (\mathbf{y}, \mathbf{H})) = E_{\mathbf{H}} \{\mathcal{I}(\mathbf{x}; (\mathbf{y}|\mathbf{H} = H))\}$

$$C = E_{\mathbf{H}} \left\{ \log\det \left(I + \frac{P}{T} \mathbf{H}\mathbf{H}^{\dagger} \right) \right\} \tag{8}$$

  - We can't take expectation inside $\log\det$.
  - P/T is the "equal power, transmit white"
- Eqn (8) is HIGHLY abused in literature. *It only applies to* $\mathbf{H}$ *being circularly symmetric Gaussian.*
  - Can ask similar (though not equivalent) question: "what capacity do we get if the input is equal-power Gaussian" for arbitrary channels.
  - Question not necessarily well posed.

[FG98] G. J. Foschini and M. J. Gans. On limits of wireless communications in a fading environment when using multiple antennas. *Wireless Personal Communications*, 6:311–335, 1998.

[MH99] T. L. Marzetta and B. M. Hochwald. Capacity of a mobile multiple-antenna communication link in rayleigh flat fading. *IEEE Trans. Inform. Theory*, 45(1):139–157, January 1999.

Proof outline

1. Start from Eqn (7) pp. 243
2. We can only choose $\mathbf{Q}$.
3. Distribution of $\mathbf{H}$ unchanged by unitary matrices: *not true in general, true for Gaussian ensembles!*
4. So, $\mathbf{Q} = \mathrm{diag}\{q_{11}, q_{22}, \ldots\}$ is a general choice.
5. Apply trace rule, $\mathrm{Tr}(\mathbf{Q}) \leq P$ implies $\sum_i q_{ii} \leq P$
6. Maximum with equality.

It's nice mathematics...

- IF each entry for $\mathbf{H}$ corresponds to an independent, flat-fading Rayleigh channel, then

$$h_{ji} = \mathcal{N}\left(0, 1/\sqrt{2}\right) + \jmath\mathcal{N}\left(0, 1/\sqrt{2}\right)$$

- Law-large-numbers: $\frac{1}{N}\sum_n^N |a|^2 \to 1$ for a complex Gaussian.

$$\boxed{\lim_{T\to\infty} C = R\log\left(1+P\right) \qquad (9)}$$

- Capacity growth is linear with respect to minimum number of transmit/receive elements.
- recall, we have already shown for T independent parallel channels, and P power constraint, the capacity is $C \leq T\log\left(1+P/T\right) \leq P\log e$ if the transmitter knows the channel.
- Ummm... but there are only T inputs to this channel, and R outputs... how can we get better growth, *when the transmitter doesn't know the channel?*

Main concepts

- MIMO exploits multipath, not mitigation...
- MIMO has linear growth wrt. $m = \min(R, T)$
  - Need full, random matrix
  - flat Rayleigh fading, with well spaced antenna elements, is a good approximation for this
  - Rich Scattering Environment
  - What is “rich?” What happens if environment is not rich?
  - a little counter-intuitive...
- Transmitter knowledge of channel not needed for linear growth:
  transmit equal power, independent signals from all elements.
- Linear growth in capacity means
  - Capacity increases without increasing power – just need more elements.
  - Many interesting theoretical results based on random matrix theory
  - Equal capacity over longer distances
  - More channels for fixed capacity (think telephone lines)

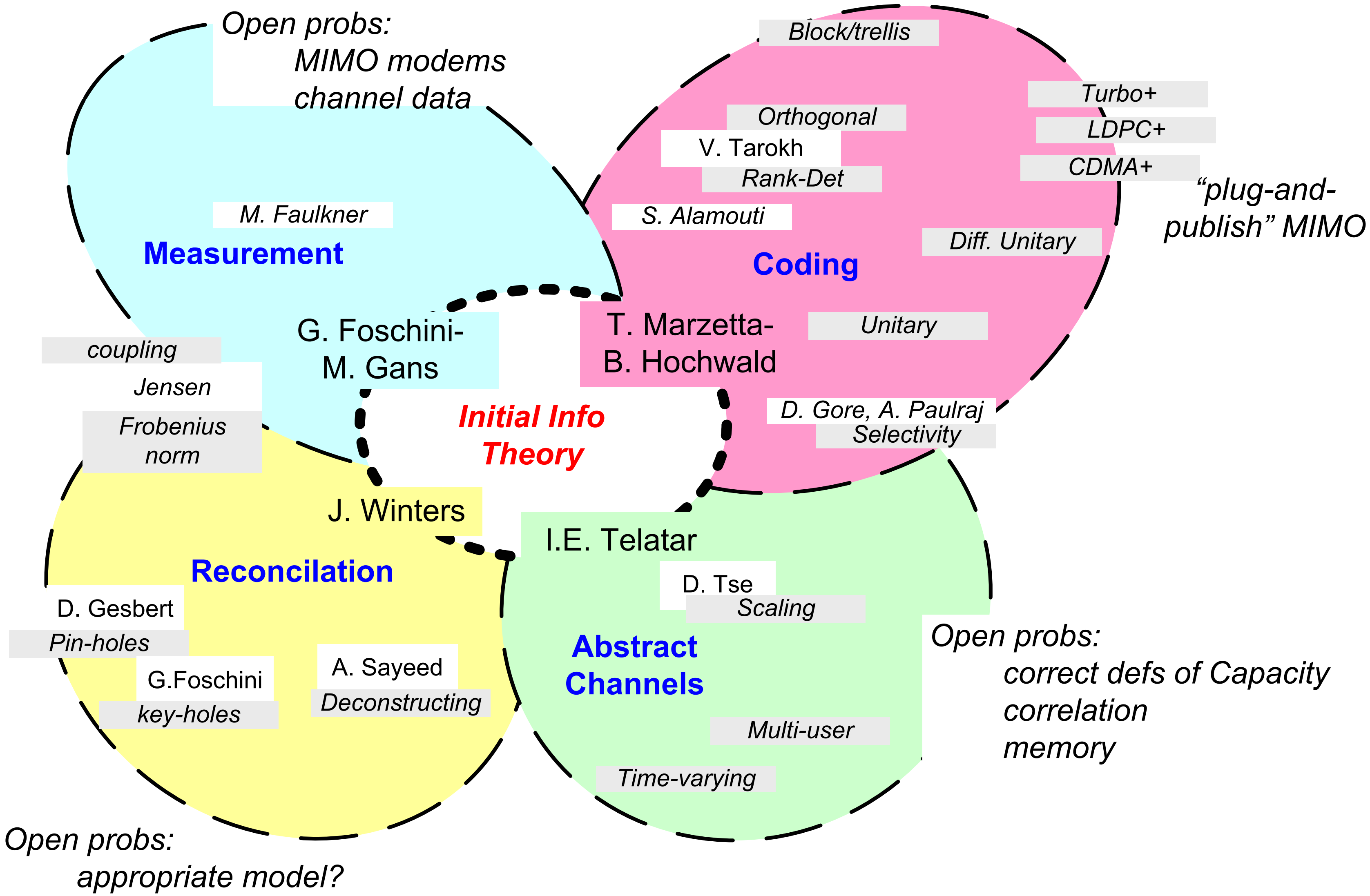
[Mül02] R. R. Müller. A random matrix model of communication via antenna arrays. *IEEE Trans. Inform. Theory*, 48(9):2495–2506, September 2002.

[CTKV02] C.-N. Chuah, D. N. C. Tse, J. M. Kahn, and R. A. Valenzuela. Capacity scaling in MIMO systems under correlated fading. *IEEE Trans. Inform. Theory*, 48(3):637–650, March 2002.

Day 2

- Extensions
- Non-ideal environments
- Coding
- Implementation

- By the end of today you should
  –


Open probs:
MIMO modems
channel data
M. Faulkner
Measurement
G. Foschini-
M. Gans
coupling
Jensen
Frobenius
norm
Block/trellis
Turbo+
Orthogonal
LDPC+
V. Tarokh
CDMA+
Rank-Det
"plug-and-
publish" MIMO
S. Alamouti
Diff. Unitary
Coding
T. Marzetta-
B. Hochwald
Unitary
D. Gore, A. Paulraj
Selectivity
Initial Info
Theory
J. Winters
I.E. Telatar
Reconcilation
D. Tse
Scaling
D. Gesbert
Pin-holes
G.Foschini
key-holes
A. Sayeed
Deconstructing
Abstract
Channels
Open probs:
correct defs of Capacity
correlation
memory
Multi-user
Time-varying
Open probs:
appropriate model?

Extensions

- Channel must be ergodic

> **DEFINITION 1 (ERGODIC).** *A source (channel) is ergodic if* ***every*** *measurable, invariant set of sequences has* ***either*** *probability one* ***or*** *probability zero.*

Ergodicity allow us to use law-of-large-numbers, and may be interpreted as saying "the average of sampled outputs equals the ensemble average" or "all possible events will (eventually) occur."

- Why is this important? What does it imply about the physical channel?

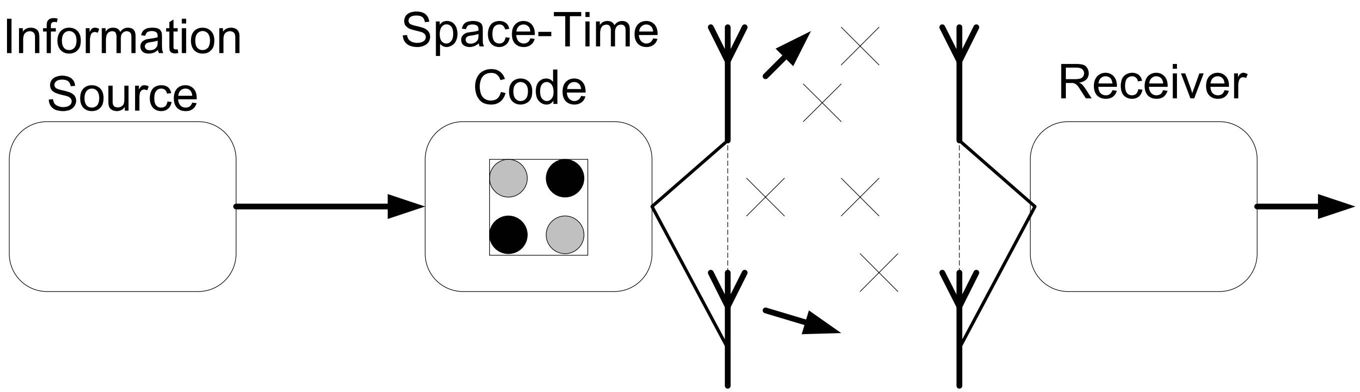
Extensions

- Capacity is a Random Variable.
  Recall, $C = E\left\{\log\det\left(I + (P/T)\mathbf{H}\mathbf{H}^{\dagger}\right)\right\}$
- Expectation is *over all possible channels*
- Random capacity, means there is probability that capacity is not achieved.

Definition 2 (Outage Capacity). *Probability that instantaneous capacity is below mean (ergodic) capacity.*

$$P_{outage} = \Pr\left[C_{instant} < E\left\{\log\det\left(I + \frac{P}{T}\mathbf{H}\mathbf{H}^{\dagger}\right)\right\}\right]$$

Using MIMO



Space-time encoding [NSC00]

- We will visit MIMO codes (briefly) later

[NSC00] A. F. Naguib, N. Seshadri, and A. R. Calderbank. Increasing data rate over wireless channels. *IEEE Signal Processing Mag.*, 17(3):76–92, May 2000.

How can we use this?

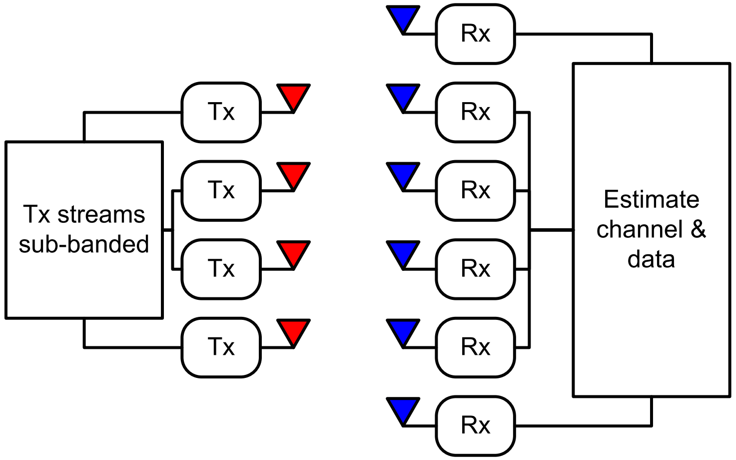
- Brute force: just send the same thing repeatedly!
- Some notation: we write $s_{t,k}$ as the symbol transmit from antenna t at symbol period k

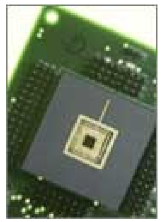
$$\left[\begin{bmatrix} s_{11} \\ s_{21} \\ \vdots \\ s_{T1} \end{bmatrix}^{k=1} \quad \begin{bmatrix} s_{12} \\ s_{22} \\ \vdots \\ s_{T2} \end{bmatrix}^{k=2} \quad \cdots \right]$$

- Every column shows the symbols transmit at time instant k. Rows are all symbols from a single transmitter, over time
  - Think of a transmitter which selects symbols of length T at each time interval.
  - Number of different "meta-codewords" is 2^T. This gives *significantly* lower probability of error.
- Channel assumed stationary

Using MIMO: The Blast experience

- Blast (and V-Blast) developed at Bell Labs to show MIMO capacity claims could be realised.
- Uncoded (simple vector data streams). $T = 8, R = 12$ Transmit power scaled $1/T$
- Train-and-estimate, then burst of data, then train ratio $20T_s : 100T_s$
- 30kHz bandwidth, achieved 780kbs indoors. 640kbs datarate
- Theory:
  - $T = 1, R = 12$, 9bit/s/Hz. conventional
  - $T = 8, R = 12$, 49bit/s/Hz. MIMO

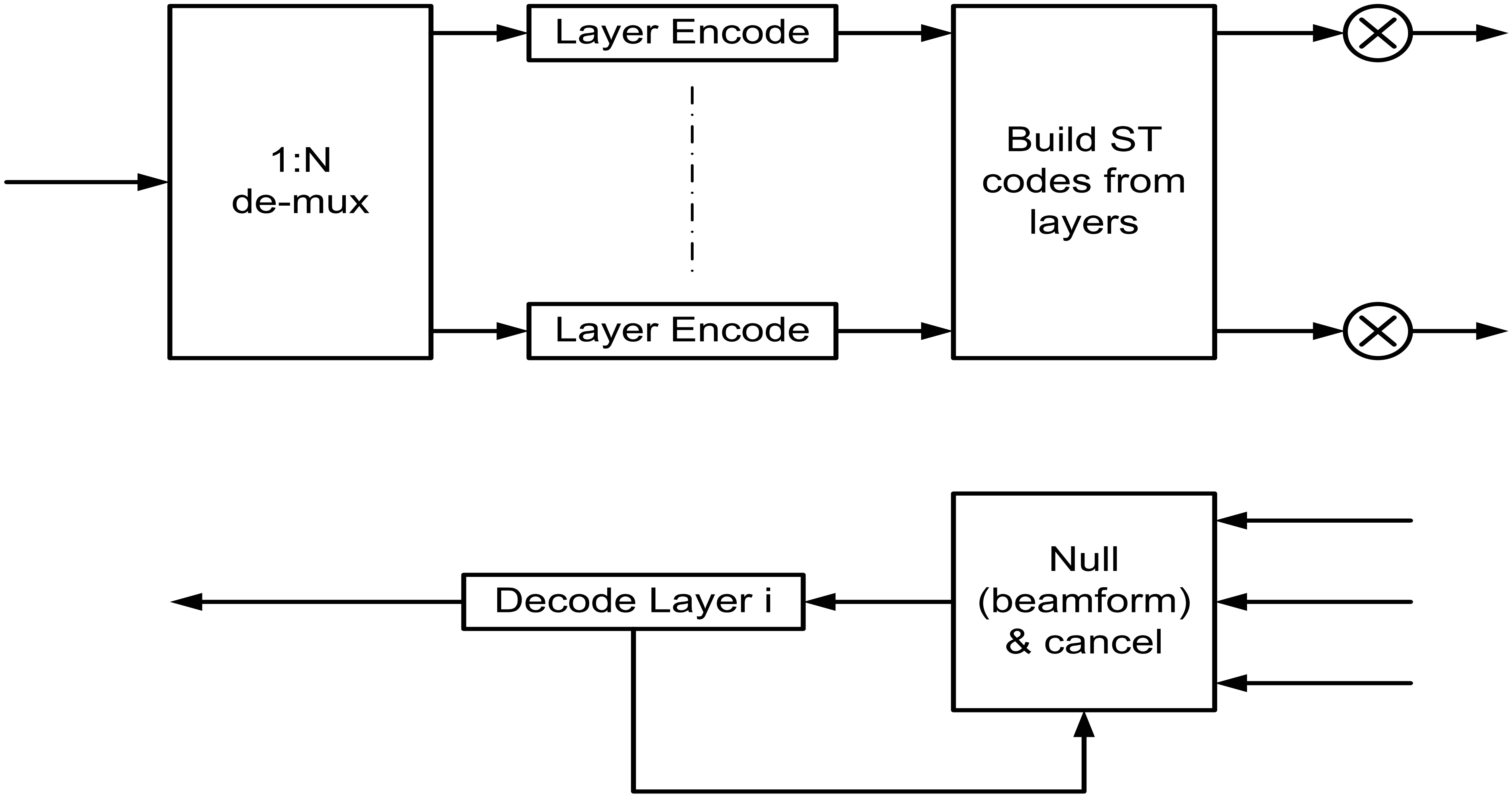




Current BLAST technology.
With perm. from Bell-Labs

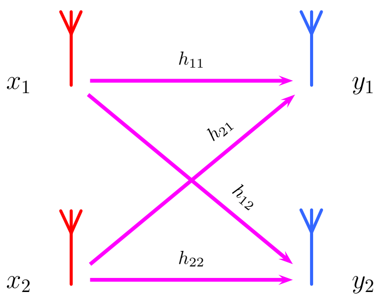
[WGFV98] P. W. Wolniansky, G. D. Golden, G. J. Foschini, and R. A. Valenzuela. V-BLAST: An architecture for realizing very high data rates over the rich-scattering wireless channel. In *URSI Int. Symp. on Signals, Systems, and Electronics, ISSSE-98*, pages 295–300, September 29 – October 2 1998.

Bell Labs lAyered Space Time

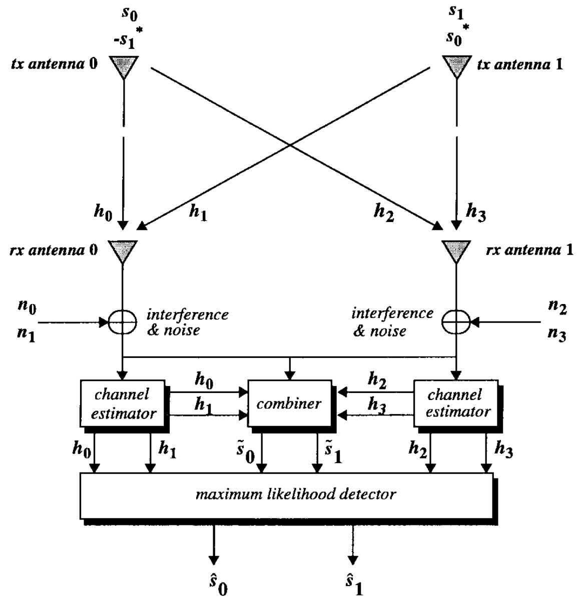


Initial steps to Space-Time Coding

- Whatever is transmitted on transmit element t appears on all receive elements $r = \{1, \ldots, R\}$. Receive signal has gain due to channel plus iid AWGN due to receiver. Initial codes for $T = R = 2$ case.



	Tx 0	Tx 1
time t	s_0	s_1
time $t + T_s$	-$\bar{s}_1$	$\bar{s}_0$



2banch diversity scheme [Ala98]

[Ala98] S. Alamouti. Space block coding: A simple transmit diversity technique for wireless communications. *IEEE J. Select. Areas Commun.*, 16:1451–1458, October 1998.

Space-Time Coding: Alamouti flavour

- Designed as simple way of generating diversity at transmitter
  - Can be used in any multi-dimensional setting *not just space-time*... although all the MIMO interest has clouded that... Not the first, but most famous.
- Received signals:

$$\left\{ \begin{bmatrix} r_0 \\ r_1 \end{bmatrix}^t \quad \begin{bmatrix} r_0 \\ r_1 \end{bmatrix}^{t+T_s} \right\} = \begin{bmatrix} h_{00} & h_{01} \\ h_{10} & h_{11} \end{bmatrix} \left\{ \begin{bmatrix} s_0 \\ s_1 \end{bmatrix} \quad \begin{bmatrix} -s_1^\dagger \\ s_0^\dagger \end{bmatrix} \right\} + \left\{ \begin{bmatrix} z_0 \\ z_1 \end{bmatrix} \quad \begin{bmatrix} z_0 \\ z_1 \end{bmatrix} \right\}$$

$$\mathbf{R} = \mathbf{HS} + \mathbf{Z}$$

- Decoder then applies channel "inverse"

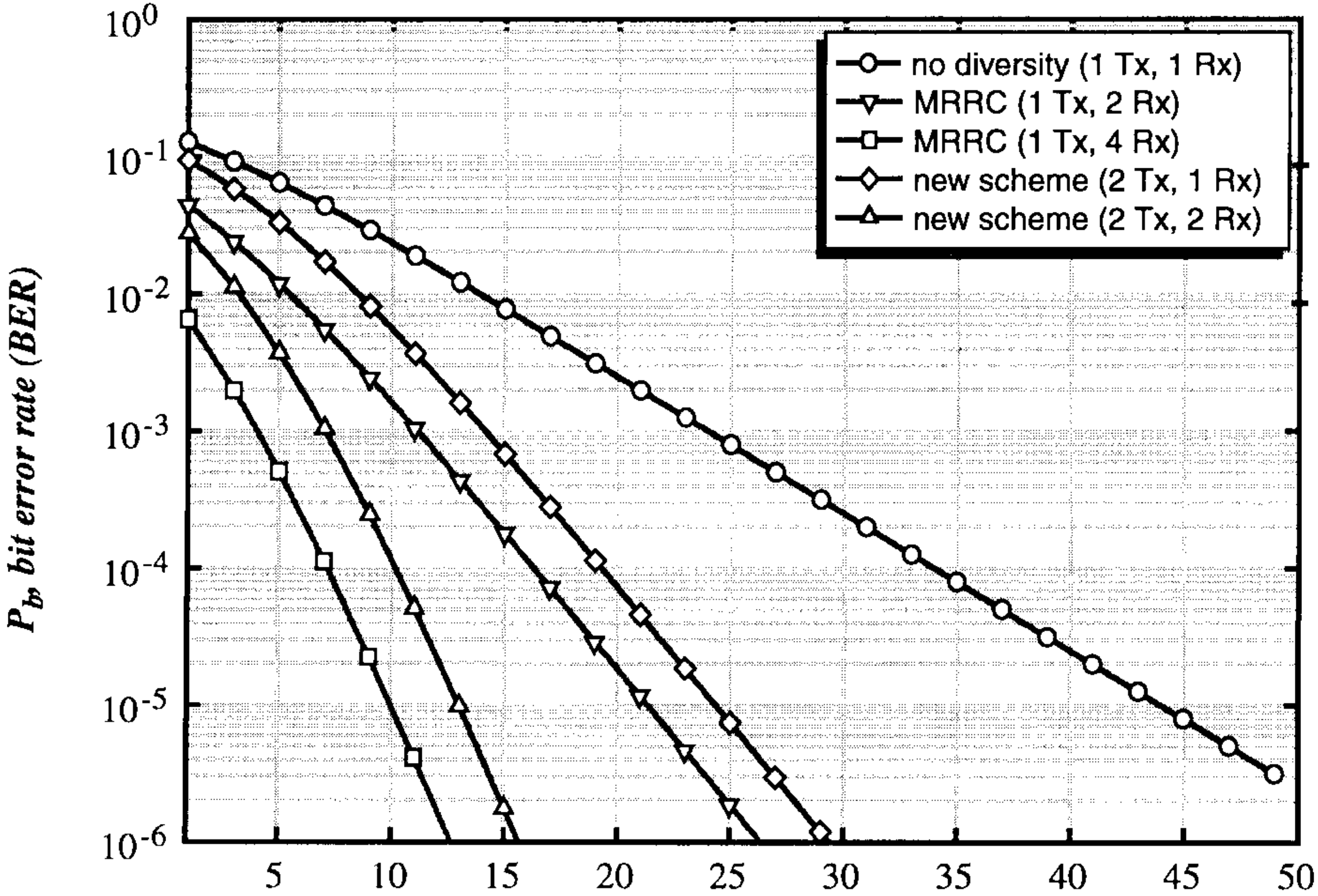
$$d_0 = \overline{h_{00}} r_{00} + h_{01} \overline{r_{01}} + \overline{h_{10}} r_{10} + h_{11} \overline{r_{11}} = s_0 \sum_{jk} |h_{jk}|^2 + w$$

$$d_1 = \overline{h_{01}} r_{00} - h_{00} \overline{r_{01}} + \overline{h_{11}} r_{10} - h_{10} \overline{r_{11}} = s_1 \sum_{jk} |h_{jk}|^2 + w$$

- How to choose correct signal? Pick the either s_1 or s_0 dependent on which is closest to d_i.

[SW93] N. Seshadri and J. H. Winters. Two signaling schemes for improving the error performance of frequency-division-duplex (FDD) transmission systems using transmitter antenna diversity. In *IEEE Vehicular Technol. Conf. (VTC 93)*, pages 508 – 511, May 18–20 1993.

Results



BER performance, coherent BPSK [Ala98, fig 4]

Alamouti problems

- Arbitrary code & decoder design
- What can we do with 3 antenna elements?
- Block codes are rarely as good as trellis codes

Tarokh, etal. Performance Criterions

- Assume random channel (nominally Rayleigh, but some potential for Ricean)
- Make use of simple trick:
  - Error probability given by ensemble average of code-word distance
  - Can be approximated by simple sum.

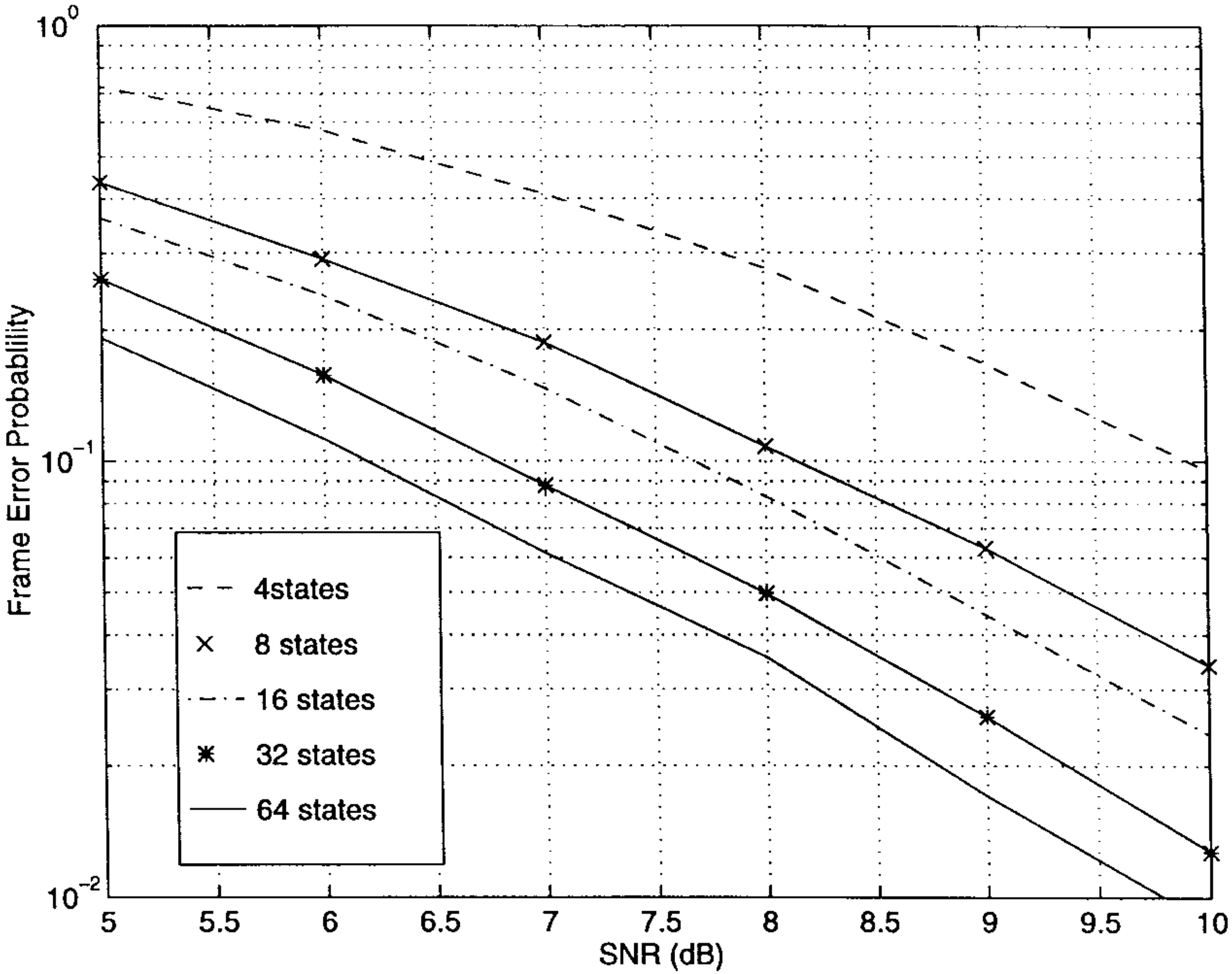
$$\Pr\left(\mathbf{c} \to \mathbf{e} | h_{ij}, i = 1, \ldots, T, j = 1, \ldots, R\right) \leq \prod_{j=1}^{R} \exp\left\{-H_{:,j} A(\mathbf{c}, \mathbf{e}) H_{:,j}^{\dagger} E_b / 4N_0\right\}$$

  - $A(\mathbf{c}, \mathbf{e})$ is the (Euclidean) code-distance matrix. This is just the (non-weighted) distance between $\mathbf{c}$ and all errors $\mathbf{e}$
  - $-H_{:,j} A(\mathbf{c}, \mathbf{e}) H_{:,j}^{\dagger}$ is the same thing, *after* H.

Tarokh, etal. Performance Criterions

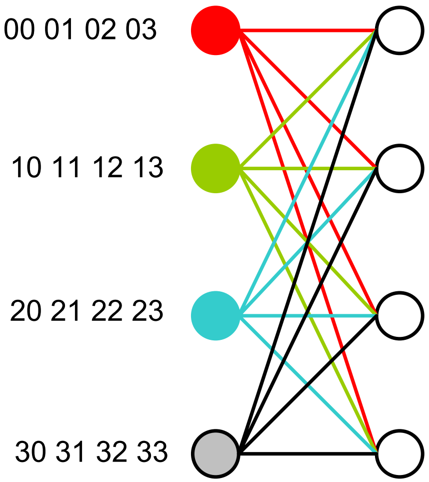
- How should we choose A?
- **Rank:** $B(\mathbf{c}, \mathbf{e}) = \sqrt{A}$ must be maximal rank, for all code words. Span all possible space. Diversity $\leq \mathrm{rank}\{B\} \min(T, R)$
- **Determinant:** min r roots, of sum of determinants of co-factors (pari-wise) for A bounds diversity. For maximum diversity, maximise the minimum roots of A.

Tarokh, etal. Results

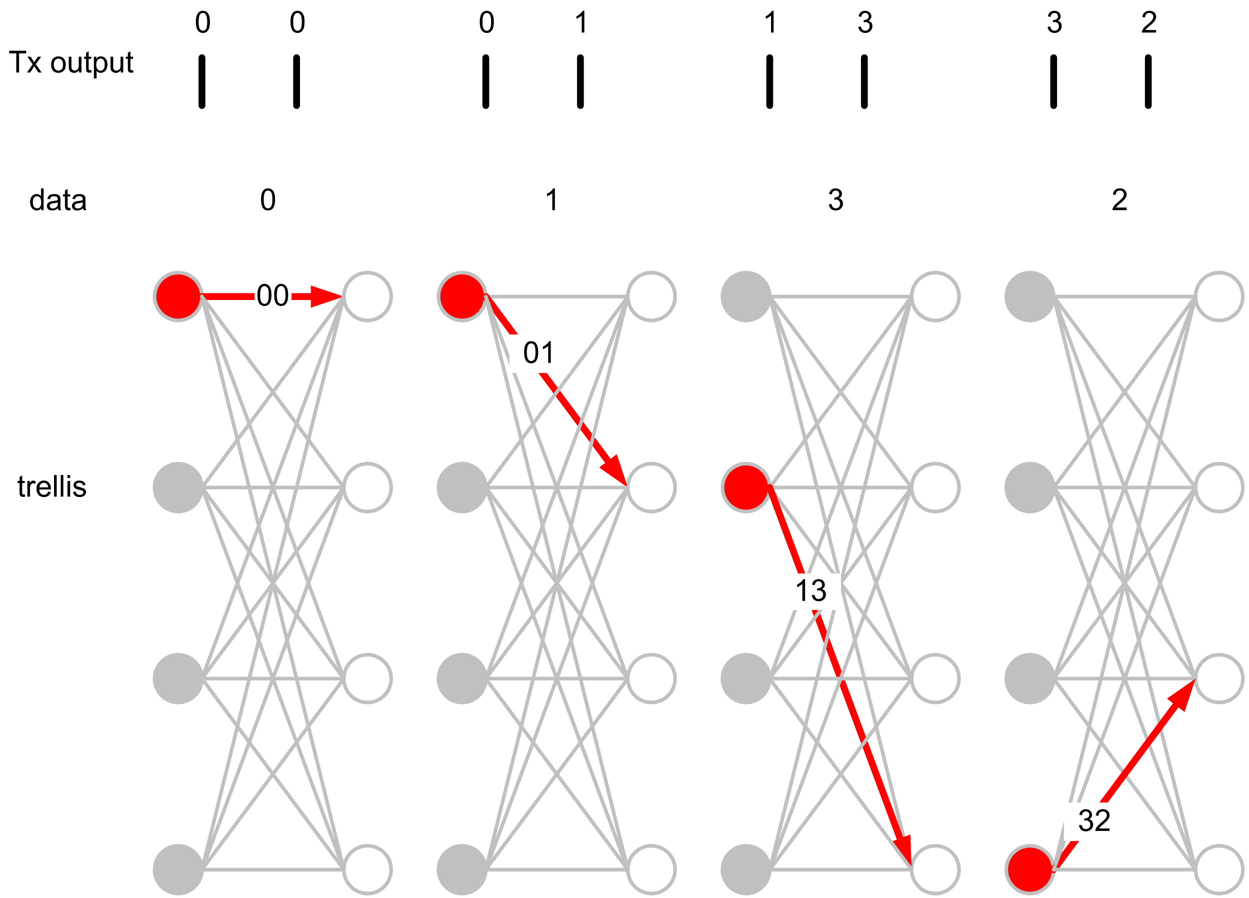


Codes for 4-PSK with rate 2 b/s/Hz that achieve diversity 4 with two receive and two transmit antennas [TSC98]. More states gives better code gain.

Example, encoding



2-transmit/receive,
4-PSK, 4state, 2b/s/Hz
[TSC98, fig 4]



[TSC98] V. Tarokh, N. Seshadri, and A. R. Calderbank. Space-time codes for high data rate wireless communication: Performance criterion and code construction. *IEEE Trans. Inform. Theory*, 44(2):744–765, March 1998.

Example, decoding

- Assumed perfect Channel state info (CSI) $h_{ij} \forall i, j$
- Receiver(s) have symbol $\{r_t^1 r_t^2 \cdots r_t^R\}$ at t
- Branch metric (for label $\{q_t^1 q_t^2 \cdots q_t^R\}$) given by

$$b = \sum_{j=1}^{R} \left| r_t^j - \sum_{i=1}^{T} h_{ij} q_t^i \right|^2$$

- Choose branch with smallest accumulated b.

Example, decoding

- Consider previous code with
  - $H = \begin{bmatrix} 1 & \jmath \\ \jmath & 1 \end{bmatrix}$
  - Constellation: 0, $\jmath$, -1, $-\jmath$
- Received symbols:

$$\begin{bmatrix} 1+\jmath \\ 1+\jmath \end{bmatrix} \quad \begin{bmatrix} 0 \\ 2\jmath \end{bmatrix} \quad \begin{bmatrix} 1+\jmath \\ -1-\jmath \end{bmatrix} \quad \begin{bmatrix} -2\jmath \\ 0 \end{bmatrix}$$

Unknown channels: unitary

- So far all codes are coherent – need full CSI at receiver.
- Codes which can perform well *without* receiver training are valuable.
- Data model

$$\mathbf{Y} = \mathbf{H}\mathbf{C}_m + \mathbf{Z}$$

- Design:
  - Choose set of unitary matrices $\mathbf{C}_m$
  - and codebook $s_m \Leftrightarrow \mathbf{C}_m$
- Detection:

$$\hat{\mathbf{C}}_m = \arg\max_{\mathbf{C}_k} \mathrm{Tr}\left(\mathbf{Y}\mathbf{C}_m^\dagger \mathbf{C}_m \mathbf{Y}^\dagger\right)$$

[HMR+00] B. M. Hochwald, T. L. Marzetta, T. J. Richardson, W. Sweldens, and R. Urbanke. Systematic design of unitary space-time constellations. *IEEE Trans. Inform. Theory*, 46(6):1962–1973, September 2000.

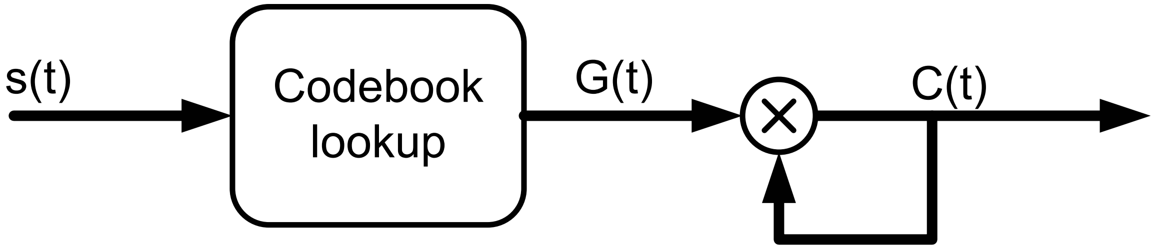
Unitary code design

- Pick a column vector with desired properties, then construct code matrix $\mathbf{S}$ by rotations in Fourier space

$$\mathbf{c} = \begin{bmatrix} c_1 \cdots c_m \end{bmatrix}$$

$$\mathbf{S} = \begin{bmatrix} \exp(\jmath 2\pi/q\mathbf{c}) & \exp(2\jmath 2\pi/q\mathbf{c}) & \exp(3\jmath 2\pi/q\mathbf{c}) & \cdots \end{bmatrix}^T$$

Unknown channels: differential



- Differential codes encode *difference* between successive symbols (NB: in MIMO this is matrix distance). Estimate data from differences, don't estimate channel.
- Received data matrix

$$\mathbf{Y}(t) = \mathbf{HC}(t) + \mathbf{Z}(t)$$

- $C(t)$ is transmit signal, $G(t)$ is a unitary code (see below)

[HS00] B. M. Hochwald and W. Sweldens. Differential unitary space-time modulation. *IEEE Trans. Commun.*, 48(12):2041 – 2052, December 2000.

Differential ST modulation

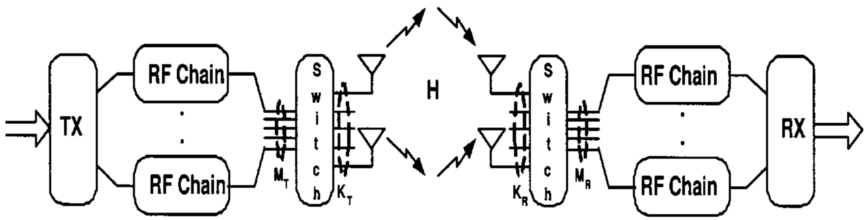
- Design:
  - Design matrix group $\mathcal{G} = \{\mathbf{G}_1, \ldots, \mathbf{G}_K\} : \mathbf{G}_i\mathbf{G}_i^\dagger = I$
  - K is number of possible symbols.
  - Choose mapping $s_k \Leftrightarrow G_k$, $k = 1, \ldots, K$
- Transmission:
  - Initial transmission is $C(0) = G_1$.
  - For each symbol $s(t)$, find the corresponding G from the codebook.
  - transmit

$$\mathbf{C}(t) = \mathbf{G}_m\mathbf{C}(t-1)$$

- Detection

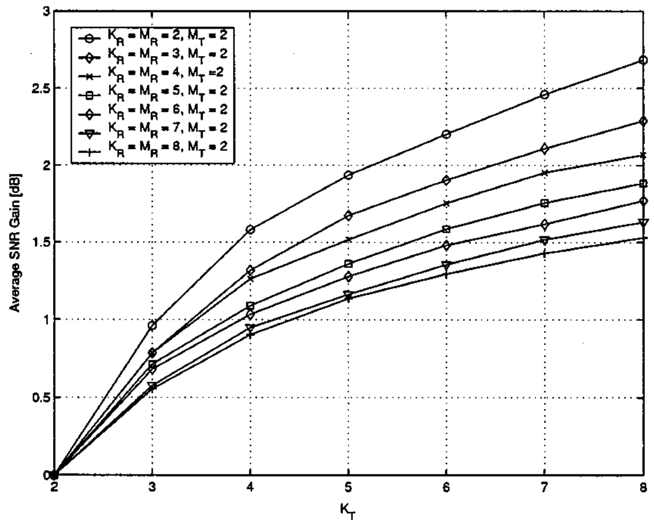
$$\hat{\mathbf{G}}_m = \arg\max_{\mathbf{G}_k} \Re\left\{\mathrm{Tr}(\mathbf{G}_k)\,\mathbf{Y}(t)^\dagger\mathbf{Y}(t-1)\right\}$$

Problem channels: selectivity



Antenna selectivity [GP02]

- Diversity: if the channel is poor, then maximise a subset of the channels.
- Full knowledge: maximise Frobenius norm
- Stat. knowledge: maximise $\det(\Sigma)$ where $\Sigma = E\{HH^{\dagger}\}$

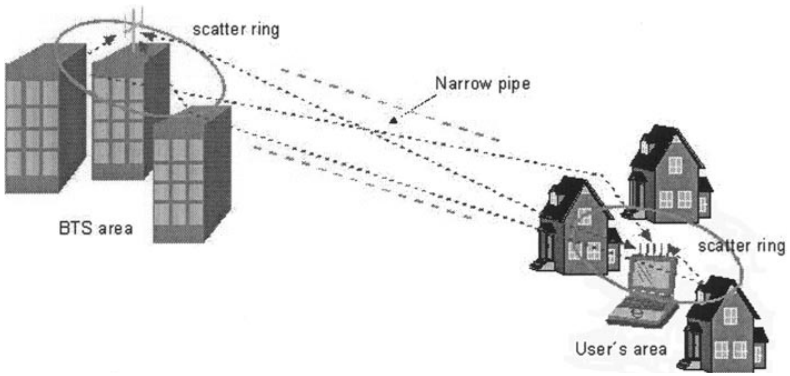


Average SNR gain with full channel knowledge, using Alamouti code [GP02]

[GP02] D. A. Gore and A. J. Paulraj. MIMO antenna subset selection with space-time coding. *IEEE Trans. Signal Processing*, 50(10):2580–2588, October 2002.

2000: Can this be real?

- Academic world essentially convinced that "linear growth" will work.
- Underlying assumption: local diversity is sufficient.
- **Gesbert etal.:** Possible to have full rank, random channel, with very low capacity. May occur when all signals pass through small region of space, even with local diversity.



Pin-hole channel [GBGP02]

Gesbert etal. triggered a rush of work relating to Kronecker models. Measurement campaigns [GAY+01a] found evidence of "key-holes"

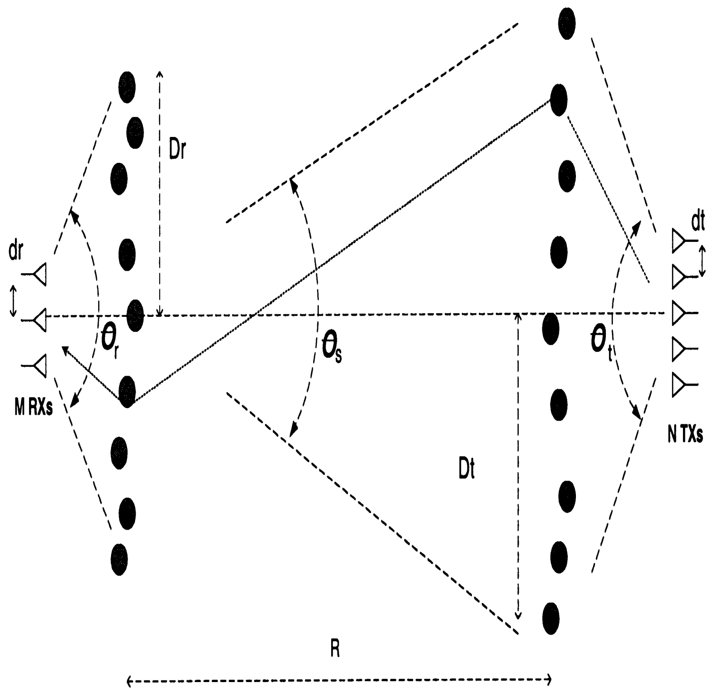
[GBGP00] D. Gesbert, H. Bölcskei, D. Gore, and A. Paulraj. MIMO wireless channels: Capacity and performance prediction. In *IEEE Global Telecommunications Conference, Globecom'00*, volume 2, pages 1083–1088, 2000.

Gesbert: scatterers as spots

- Assume equal power on all scatterers, gives a $1/\sqrt{N_s}$ normalisation, where N_s = number of scatterers.
- Arrays give correlation at each end of the link. Scatterers live on rings (defined abstractly, can be considered as power limit) and produce "virtual arrays"
- Gives overall correlation model:

$$H = \frac{1}{\sqrt{S}} R^{1/2}_{\text{rx}:\theta_r} X^{\text{iid}}_1 R^{1/2}_{\text{s}:2D/S} X^{\text{iid}}_2 R^{1/2}_{\text{rx}:\theta_r}$$

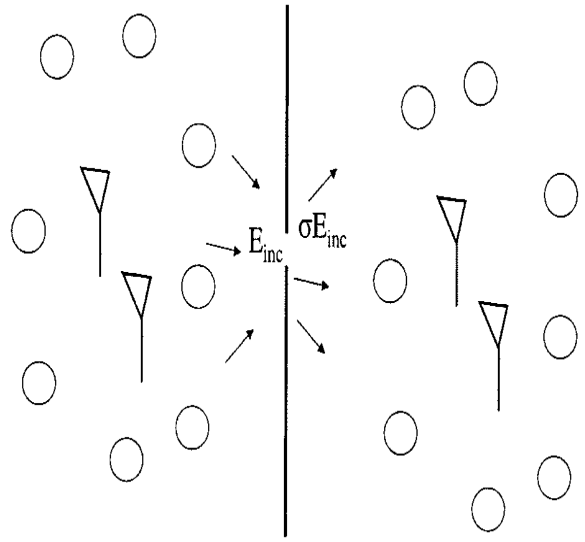
- centre correlation due to "focussing" of scattering rings. ***Weakness:*** virtual arrays.



Virtual scattering array [GBGP02]

Pin-holes vs key-holes.. debate rages

- **Key-holes and pin-holes are not the same!**
- Key-hole caused by inherent-low rank channel
- Pin-hole caused by focussing of scatter
- In the limit, key-hole = pin-hole.



Key-hole concept [CFGV02]

Key-holes & pin-holes both demonstrate local-diversity is not enough. Although some evidence of key-holes exists, they are hard to produce.

[CFGV02] D. Chizhik, G. Foschini, M. Gans, and R. Valenzuela. Keyholes, correlations, and capacities of multielement transmit and receive antennas. *IEEE Trans. Commun.*, 1(2):361–392, April 2002.

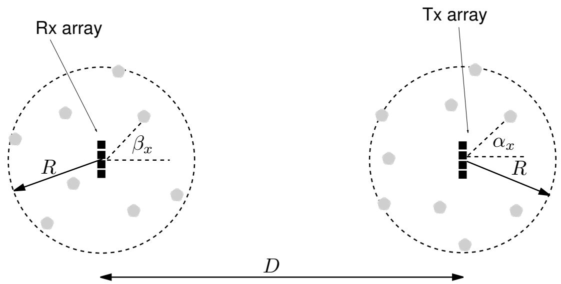
[GAY+01] M. Gans, N. Amitay, Y. Yeh, H. Xu, R. Valenzuela, T. Sizer, R. Storz, D. Taylor, W. MacDonald, C. Tran, and A. Adamiecki. BLAST system capacity measurements at 2.44GHz in suburban outdoor environment. In *VTC 2001 Spring, IEEE Vehicular Technol. Conf. (VTC 01)*, volume 1, pages 288–292, Rhodes, Greece, May 6–9 2001.

Local ring model

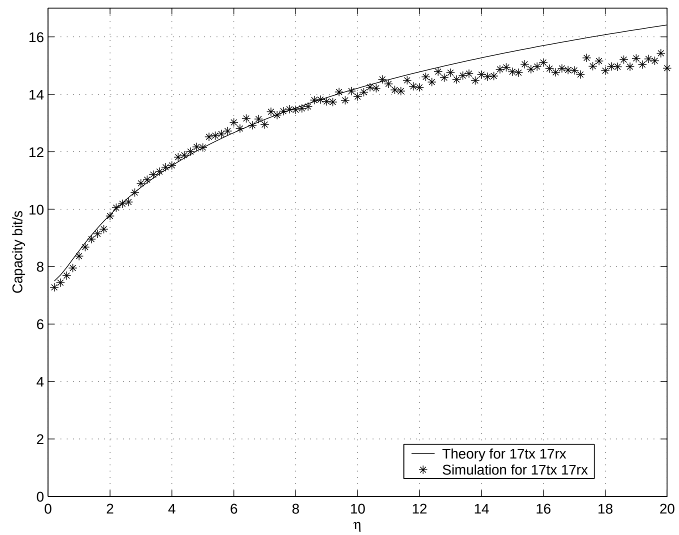
- Can extend Gesbert model [GBGP02] to allow for disks of scatterers (scatterers lie anywhere within rings
- pin-hole essentially due to ratio: $\epsilon = \frac{2\pi R}{\lambda D}$

$$C' \leq \log_2 (1 + PR) + R\kappa \log_2 \left(1 + P\epsilon^2\right)$$

κ constant.



Local scatter rings [HF02]

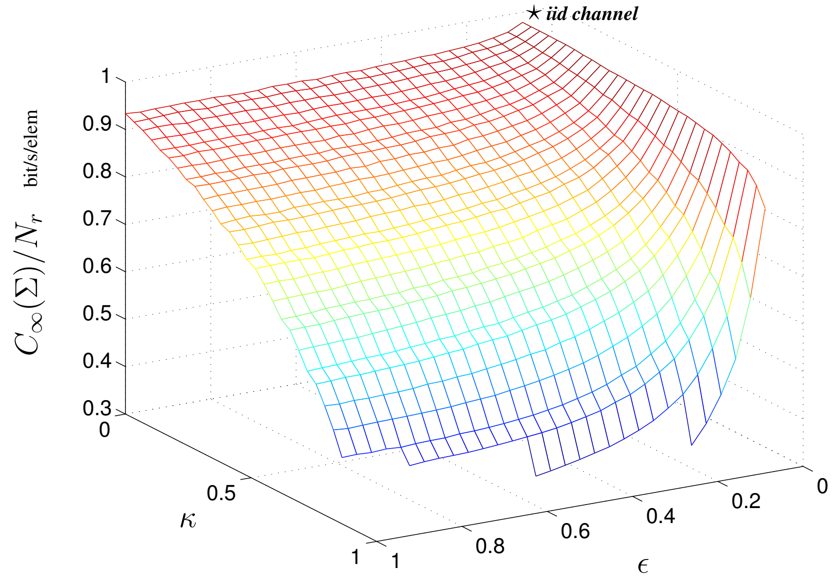


Local scatter rings [HF02]

[HF02] L. W. Hanlen and M. Fu. Multiple antenna wireless communication systems: Limits to capacity growth. In *IEEE Wireless Communication and Networking Conference, WCNC'02*, pages 172–176, Orlando, USA, March 17 – 21 2002.

Current state of the art (theory)

- Initial work sparked renewed interest in channel modelling.
  - Similarly to fading models, MIMO channel models come in many flavours, depending on the context.
- Basic result:
  - Linear growth always happens, just at different rates *if random matrix assumed*.
- Work is commencing on continuous space models



Capacity surface [HG03]

[YO02] K. Yu and B. Ottersten. Models for MIMO propagation channels: a review. *Wireless Communications and Mobile Computing*, 2:653–666, November 2002.

[CRT01] N. Chiurtu, B. Rimoldi, and E. Telatar. Dense multiple antenna systems. In *IEEE Inform. Theory Workshop*, pages 108–109, Cairns, Australia, September 2–7 2001.

Current state of the art (practical)

- Several modems either in test mode (eg. Bell Labs) or in production (eg. Uni Newcastle, CSIRO)
- Channel test results in indoor, outdoor-urban becoming cohesive
- Coding?
  - “plug-and-publish” MIMO currently big
  - New developments? Multi-user, iterative, enhanced channel-knowledge
- Channel models?
  - Lots!

References

[Ala98] S. Alamouti. Space block coding: A simple transmit diversity technique for wireless communications. *IEEE Journal on Selected Areas in Communications*, 16:1451–1458, October 1998.

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Overview of Wireless Communications
Radio Interface Protocols

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Overview-Mobile Networking(1) [1, 2]

- Radio Resource Management
  - Power Control
  - Handover Control
  - Congestion Control
  - Measurement of Load
  - Admission Control

- Radio Resource Control Protocol
  - RRC Architecture
  - RRC Functions

Overview-Mobile Networking (2)

- Medium Access Control
  - MAC Architecture for 3GPP
  - MAC Functions
  - Mapping between logical and Transport Channels
  - example
- Radio Link Control Protocol
  - RLC Architecture
  - RLC Functions
  - Example

Overview

- **Radio Resource Management**
- Radio Resource Control Protocol
- Medium Access Control
- Radio Link Control Protocol

Radio Resource Management

- Responsible for efficient use of the air interface

- RRM is needed to guarantee Quality of Service (QoS)

- Maximises capacity of system and coverage of network

- Algorithms include
  - Handovers
  - Power Control
  - Admission Control
  - Load Control
  - Packet Scheduling Functions

- Power Control is in mobile and Basestation

- Other functions are located at Network Controller (Software)

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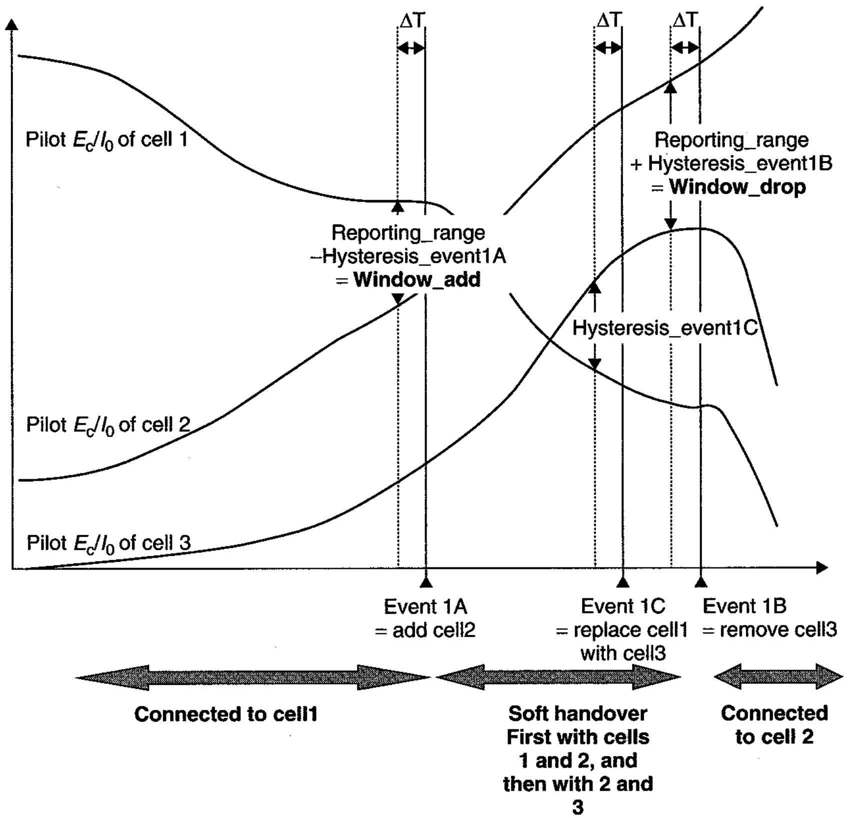
Handovers

- Allows a user to change from one cell to another
- A fundamental and important part of any cellular system
- In DS/CDMA systems a “make before break” (soft handover) is possible
- Dropped calls due to handovers is an important parameter operators study

Soft Handover

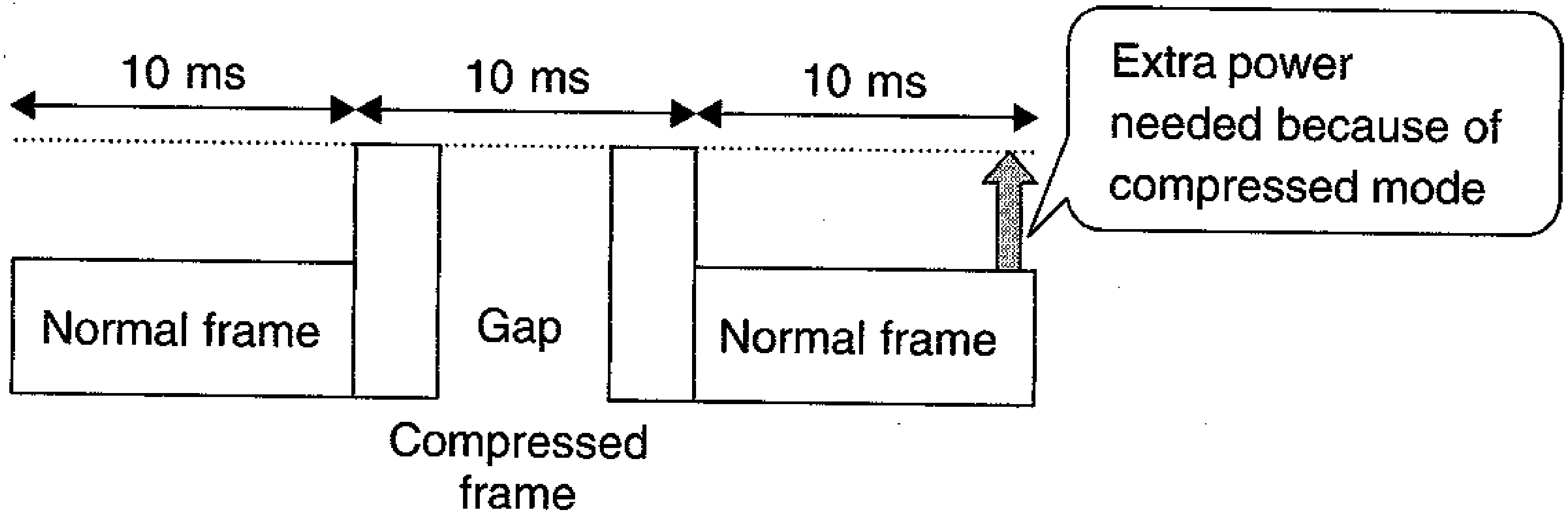
- Terminals constantly scan for other cells
- Searches for synchronisation code that all basestations send
- The terminal can identify cells with a secondary sync. code
- Terminal Reports measurement info to Radio Network Controller (RNC)
- RNC sends update command to terminal
- Gains from Soft Handover include
  - ∗ Macro diversity gain - over slow fading
  - ∗ Micro diversity gain - over fast fading
  - ∗ Downlink load sharing - Improved coverage
- Soft Handover Overhead in extra resources (no free lunch!)



Intersystem Handover (1)

- Handovers between WCDMA and GSM system possible
- This allows coverage extension by falling back to GSM
- This allows capacity extension by switching to WCDMA
- The 3GPP Standard has a gap in transmission so GSM can check the channel
- This is called Compressed mode as WCDMA sends more data before gap to allow gap Picture page 237
- Interfrequency handovers are possible as carriers have more than one band
- Most companies have three frequency bands in up/downlinks
  - Network commands terminal to start inter-frequency search
  - Terminal finds new carrier and informs RNC
  - RNC commands terminal to perform handover

Intersystem Handover (2)



- A free “Gap” is needed to allow other standard to connect to system
- To prevent reduced data rate transmission, data is sent with lower processing gain
- Result is that more power is needed to send the compressed frames

Power Control

- Fast Power Control is provided in 3GPP (1.5kHz) in up/downlinks
- GSM has only slow power control (2Hz), IS-95 (800Hz)
- For mobile user speeds less than 50km/hr gains of 2-6dB are possible
- For speeds greater than 50 km/hr there is a loss in using fast power control
- At low speeds the power control can compensate for fading
- less diversity means sharper fading and inaccurate compensation

Power Control in Soft Handover

- Power drifting in the basestation powers in the downlink
  - Tx Power of one BS could rise and another fall (power drifting)
  - Due to fast power control in the downlink
  - Solution is to restrict power control range of terminal, or have system control monitor multiple basestations

- Reliable detection of the uplink power control commands in the terminal
  - Improved by setting a higher power for the control (signalling) channel

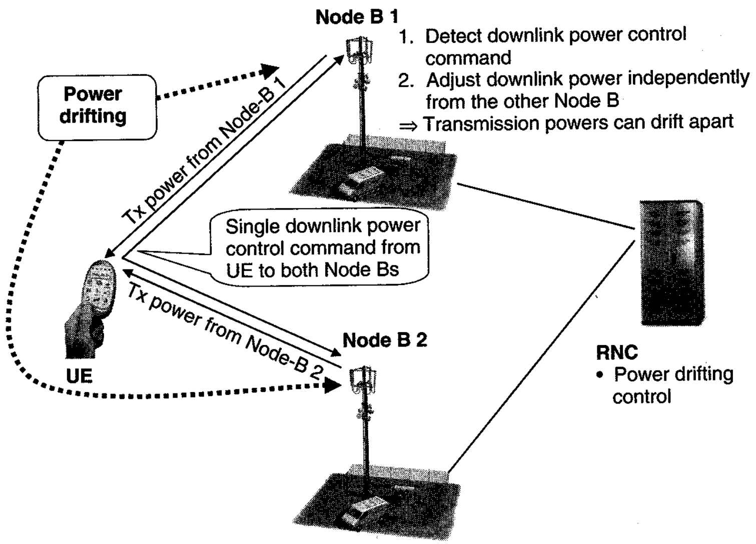


Figure 9.6. Downlink power drifting in soft handover

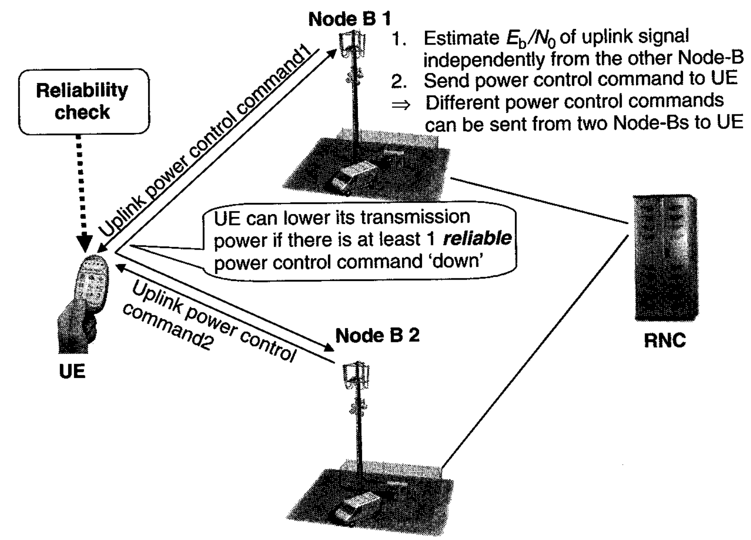
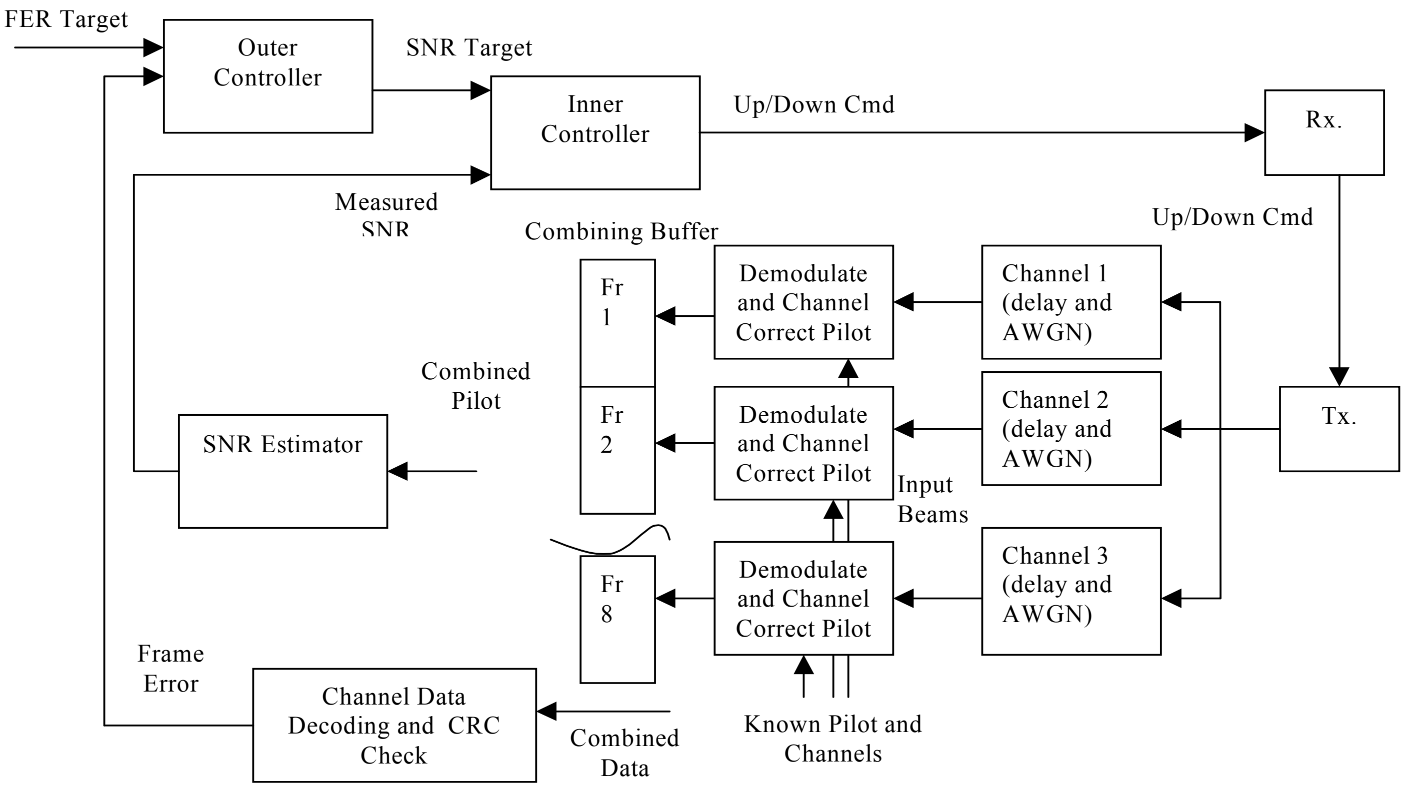


Figure 9.7. Reliability check of the uplink power control commands in UE in soft handover

Power Control Overview

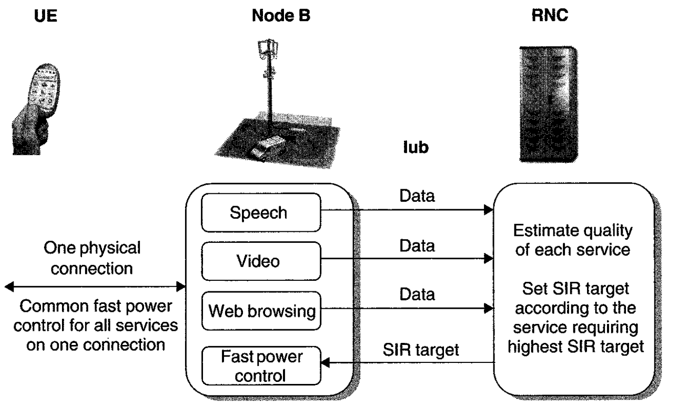


- Two loops, inner and outer
- Outer loop tries to maintain a certain FER
- Inner loop tries to maintain a given SIR (from outer loop)

Outer Loop Power Control

The outer controller's algorithm is summarised in the following pseudo code

```
 if frame_error = 0
  snr_target = snr_target  -  outer_down_step
else
  if number of up commands in pc_cmd_mem < up_max
    snr_target = snr_target + outer_up_step
  else
    do nothing
  end
end
```



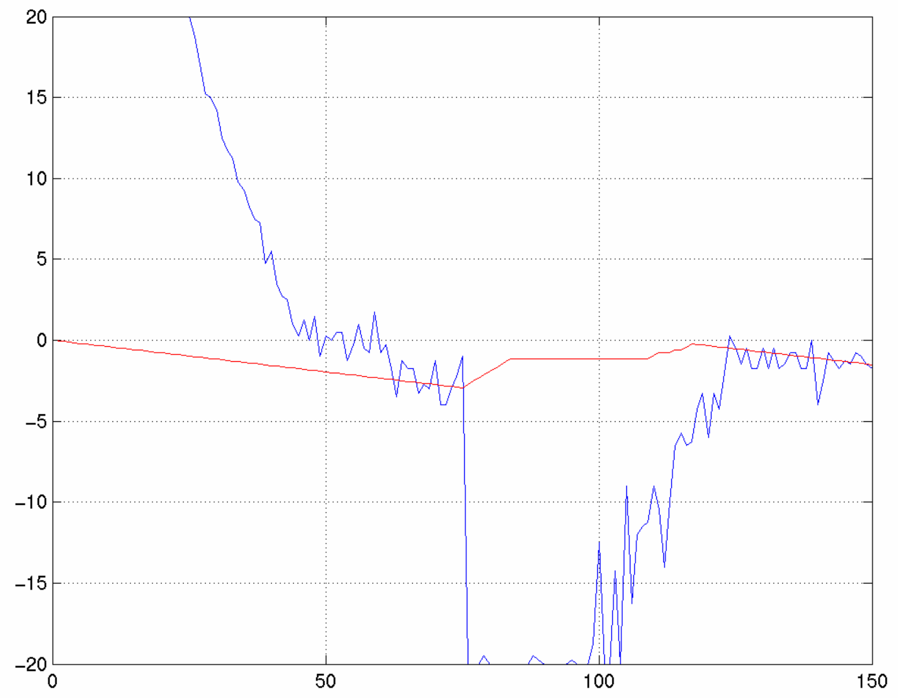
Inner Loop Power Control

The inner controller pseudo code is as follows

```
snr_diff = snr_est  snr_target
snr_diff_comp = snr_diff + sum(pc_cmd_mem)
if abs(snr_diff_comp) < threshold
 if snr_diff_comp <=0
   pc_cmd = small_step
 else
   pc_cmd = -small_step
 end
else
 if snr_diff_comp <=0
   pc_cmd = large_step
 else
   pc_cmd = -large_step
 end
end;
save pc_cmd_mem in a buffer
```

Power Control Results

- x-axis Frame number, on y-axis is the SNR in decibels
- Power level set 15dB higher than the SNR Target
- after 75 symbols the channel reduced the received power by 30dB (simulate fading)
- System had a 12 frame delay as it is simulating a Satellite system
- Frame errors simulated with no frame errors above 0dB and full frame errors below -5dB
- SNR Target (Red Line) - relatively flat, Estimated SNR (Blue Line)



Admission Control Principle

- If air interface is overloaded coverage area is reduced and QoS cannot be guaranteed
- Admission control checks that a new user will not disrupt the network BEFORE admitting them (better to disallow call than compromise whole network)
- Admission control located in RNC, where load info. for several cells can be used
- Limits of admission control are set by radio network planning
- Several algorithms,
  - total power received by basestation for uplink,
  - total downlink power is used for downlink

Admission Control Strategy

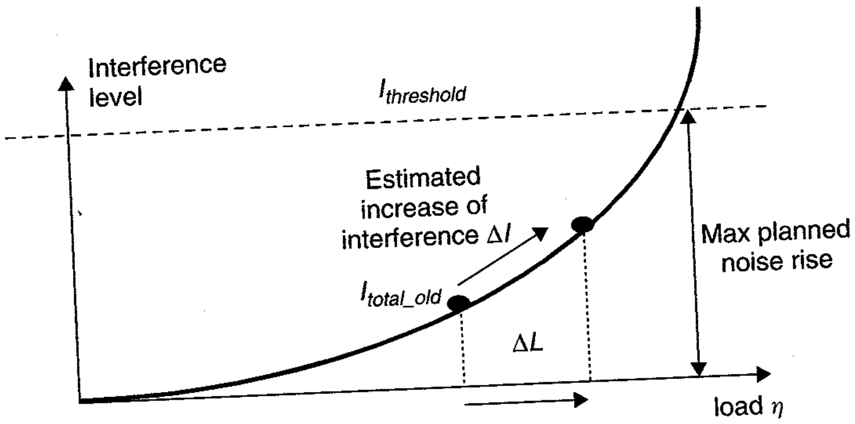
- New terminal is by the uplink if the total interference is higher than the threshold value

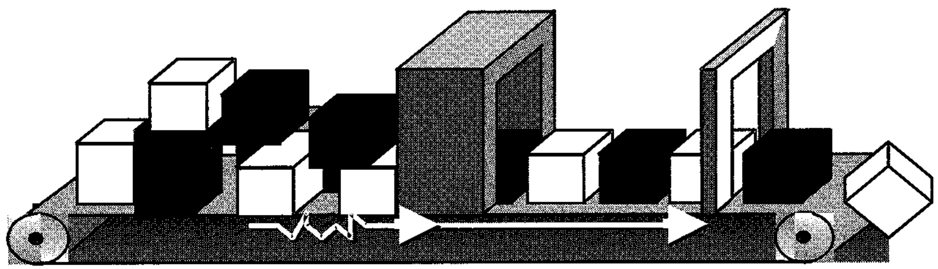
- $I_{\text{total\_old}} + \Delta < I_{\text{threshold}}$

- Methods of computing noise rise are derivative method or integral method

- Derivative: $\Delta I \approx \frac{dI_{\text{total}}}{1-\eta}\Delta L$

- Integral: $\Delta I = \frac{I_{\text{total}}}{1-\eta-\Delta L}\Delta L$





Packet ordering

Load (Congestion) Control

- Important function of Radio Resource Mgmt (RRM) is to ensure system remains stable

- If overload occurs the system needs to be returned quickly to a stable state

- The possible load control actions in order to reduce load are:-
  - Deny downlink power-up commands received from the terminal (fast)
  - Reduce the uplink E_b/N_0 target used by the uplink fast power control (fast)
  - Reduce the throughput of the packet data traffic (slower)
  - Handover to another WCDMA carrier (slower)
  - Handover to GSM (slower)
  - Decrease bit rates of real-time terminals using Adaptive multirate speech coding (slower)
  - Drop low priority calls in a controlled fashion (slow)

Overview

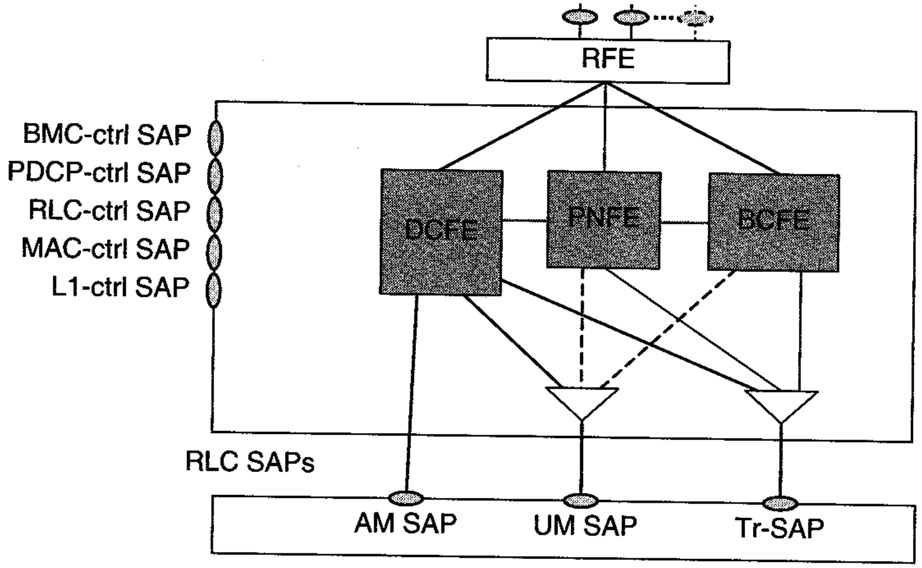
- Radio Resource Management
- **Radio Resource Control Protocol**
- Medium Access Control
- Radio Link Control Protocol

RRC Protocol

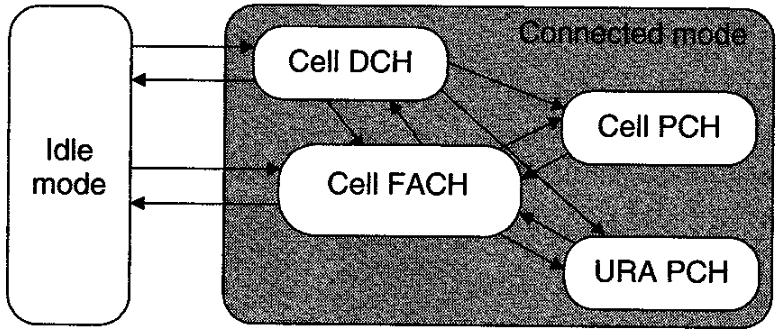
- Majority of signalling between terminal and System is RRC messages
- Carries all messages to setup, modify and release layer 2/1 protocols
- RRC messages carry in payload all higher layer signaling
- mobility of terminals is controlled by the RRC (measurements, handover, cell updates)

RRC Logical Architecture

- Dedicated Control Function Entity - handles all functions and signalling specific to a terminal
- Paging and Notification Function Entity
- Broadcast control function entity - handles broadcast System Information

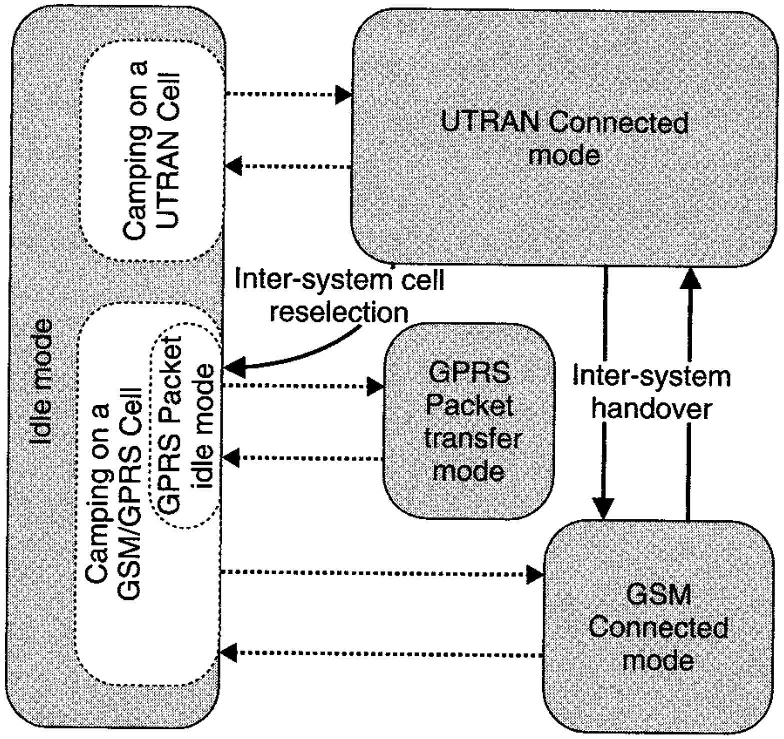


RRC Service States



- Operational Modes - Idle Mode and Connected Mode

- Further service states for exact usage
  - Cell-DCH - dedicated physical channel is allocated to terminal
  - Cell-FACH - Random channels are allocated to terminal
  - Cell-PCH - Terminal can only be reached via the paging channel



RRC Functions

- Broadcast of system info, Paging, Initial cell selection
- establishment, maintenance and release of an RRC connection
- Control of radio bearers, and channels
- control of security functions
- Integrity protection of signaling messages
- terminal measurement reporting
- RRC connection mobility functions
- support of outer power control
- cell broadcast services

Overview

- Radio Resource Management
- Radio Resource Control Protocol
- **Medium Access Control**
- Radio Link Control Protocol

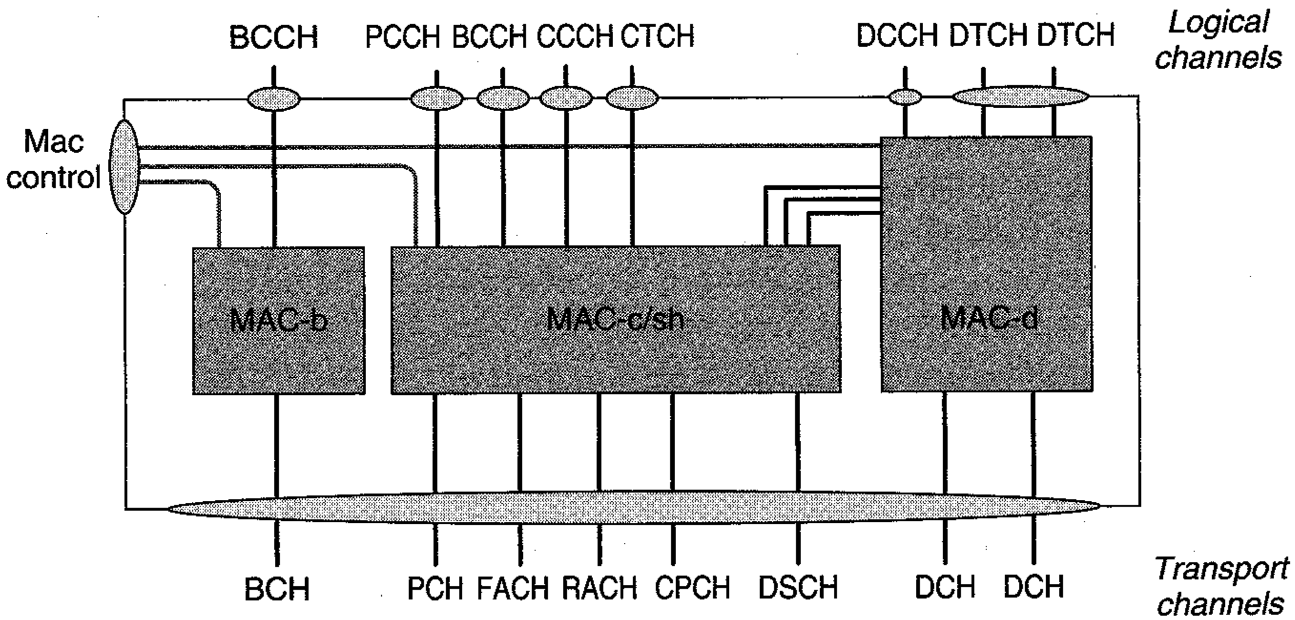
MAC Overview

- MAC layer and logical channels are mapped to transport channels

- MAC layer is responsible for selecting the appropriate transport format

- Transport format selected based on source rates and admission control of each connection

MAC Architecture for 3GPP



- MAC-b handles broadcast channel
- MAC-c/sh handles common channels and shared channels
- MAC-d handles dedicated channels allocated to terminals in connected mode

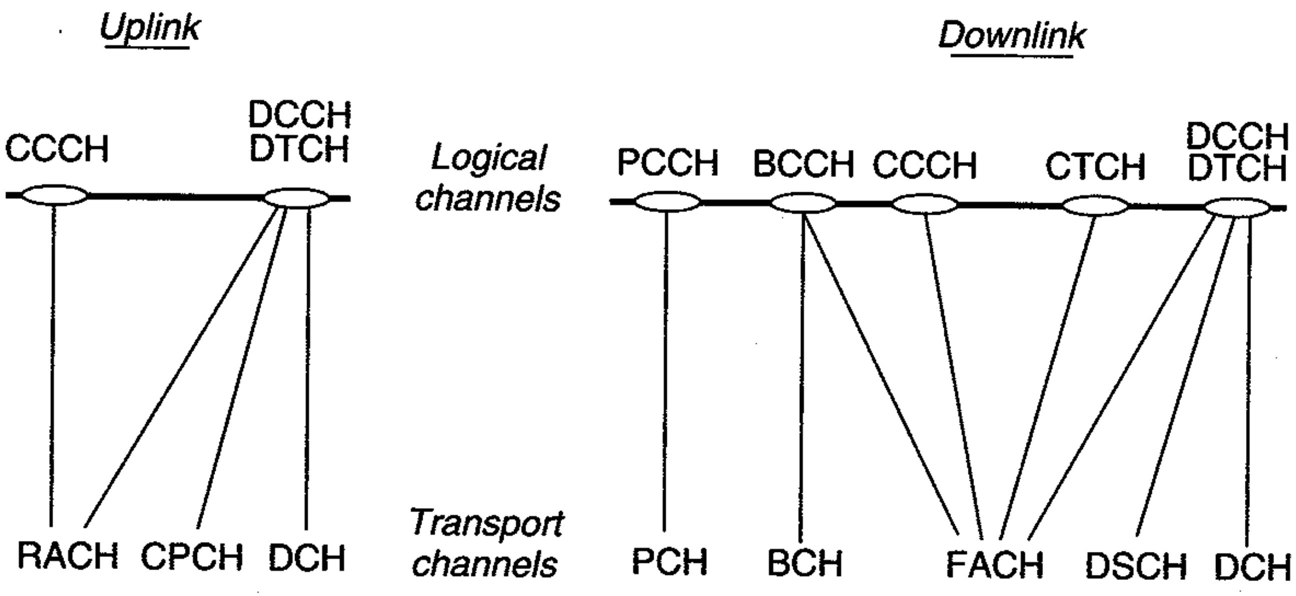
MAC Functions

- Mapping between logical and transport channels
- Selection of appropriate transport format (selecting right size package for parcel)
- Priority handling of data flows
- Identification of terminals on common transport channels
- Multiplexing of packets into transport blocks
- Traffic volume monitoring

Logical Channels

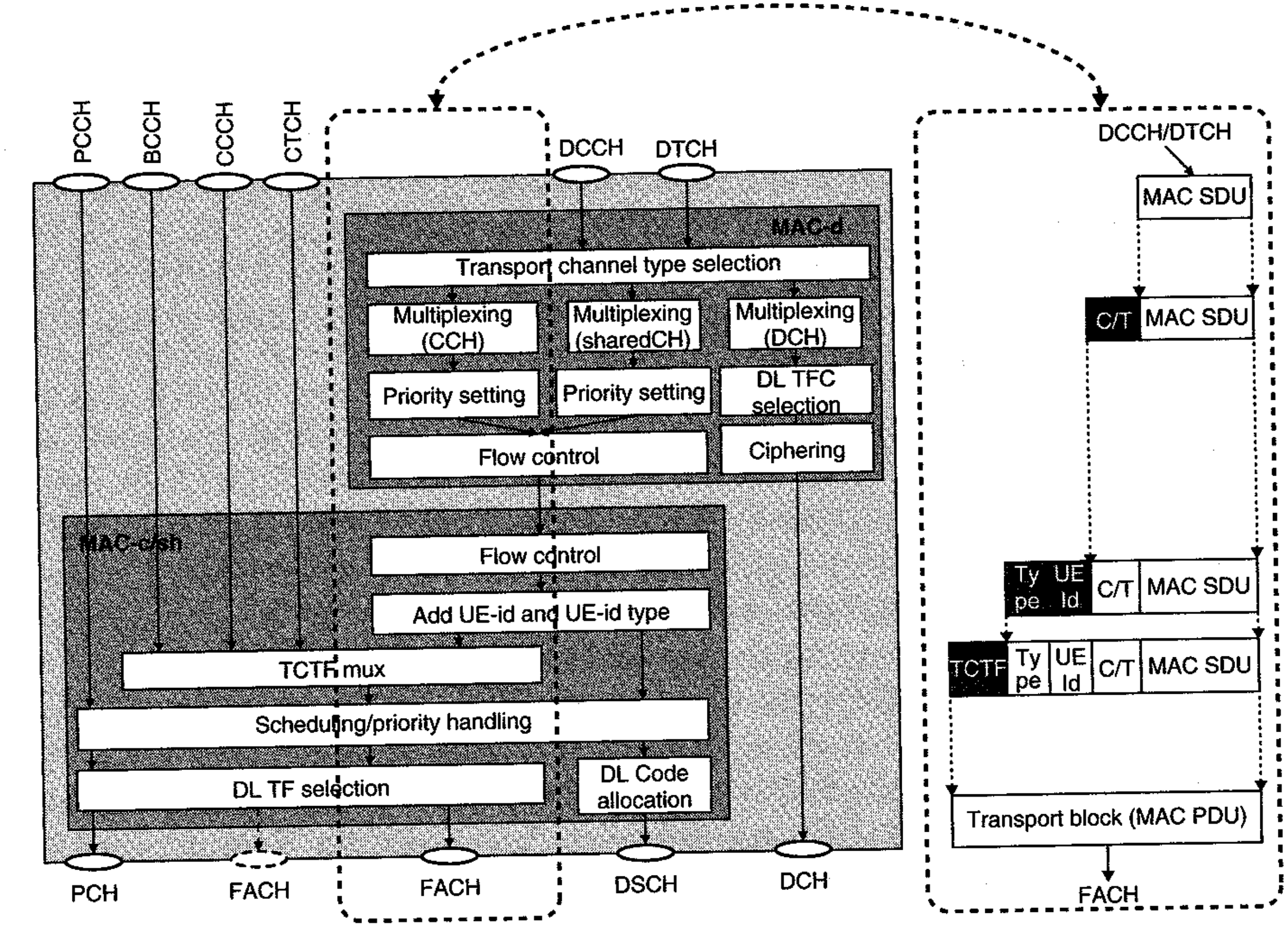
- The logical breakdown of the channels
- Broadcast Channel
- Paging Channel
- Dedicated Control Channel
- Common Control Channel
- Dedicated Traffic Channel
- Common Traffic Channel
- Transport channels are physically implemented to carry the information

Mapping between Logical and Transport Channels



- Separation of Physical (Transport) Channels from Logical Channels is important to allow protocols to operate on data in logical groups
- Separation is also part of ISO Stack model

MAC- Example



PCCH
BCCH
CCCH
CTCH
DCCH
DTCH
MAC-d
Transport channel type selection
Multiplexing (CCH)
Multiplexing (sharedCH)
Multiplexing (DCH)
Priority setting
Priority setting
DL TFC selection
Flow control
Ciphering
MAC-c/sh
Flow control
Add UE-id and UE-id type
TCTF mux
Scheduling/priority handling
DL TF selection
DL Code allocation
PCH
FACH
FACH
DSCH
DCH
DCCH/DTCH
MAC SDU
C/T
MAC SDU
Ty pe
UE Id
C/T
MAC SDU
TCTF
Ty pe
UE Id
C/T
MAC SDU
Transport block (MAC PDU)
FACH

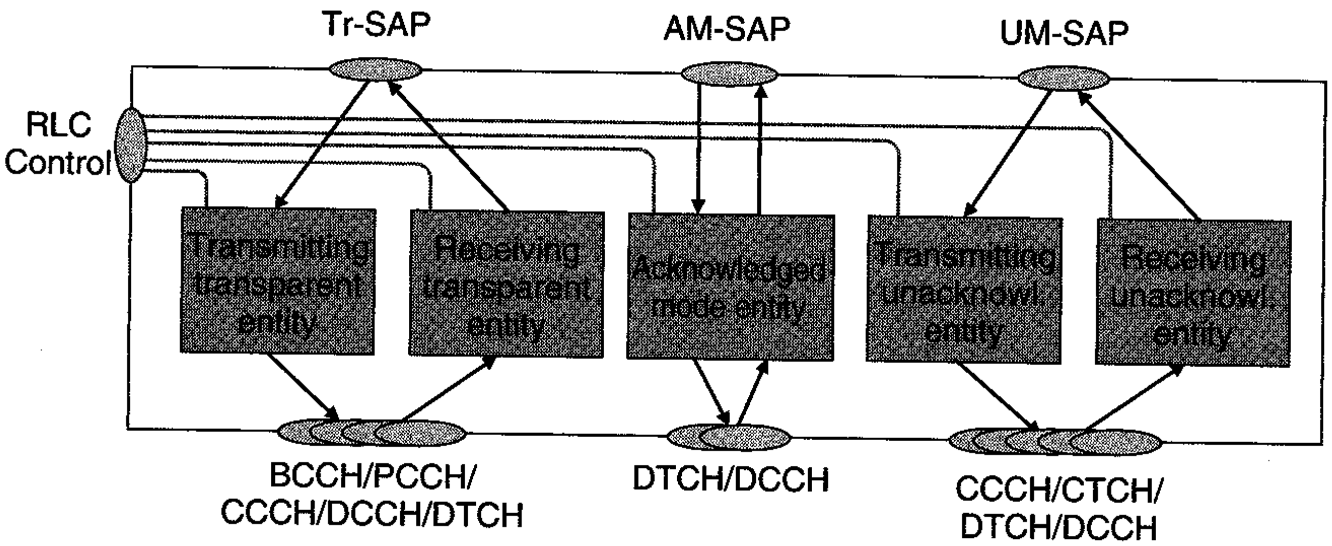
Overview

- Radio Resource Management
- Radio Resource Control Protocol
- Medium Access Control
- **Radio Link Control Protocol**

RLC Protocol

- Provides Segmentation and retransmission services
- Three possible modes:-
  - Transparent Mode - no protocol overhead is added to higher layer data
  - Unacknowledged Mode - no retransmission protocol is in use
  - Acknowledged Mode - an automatic repeat request (ARQ) mechanism is used

RLC Architecture

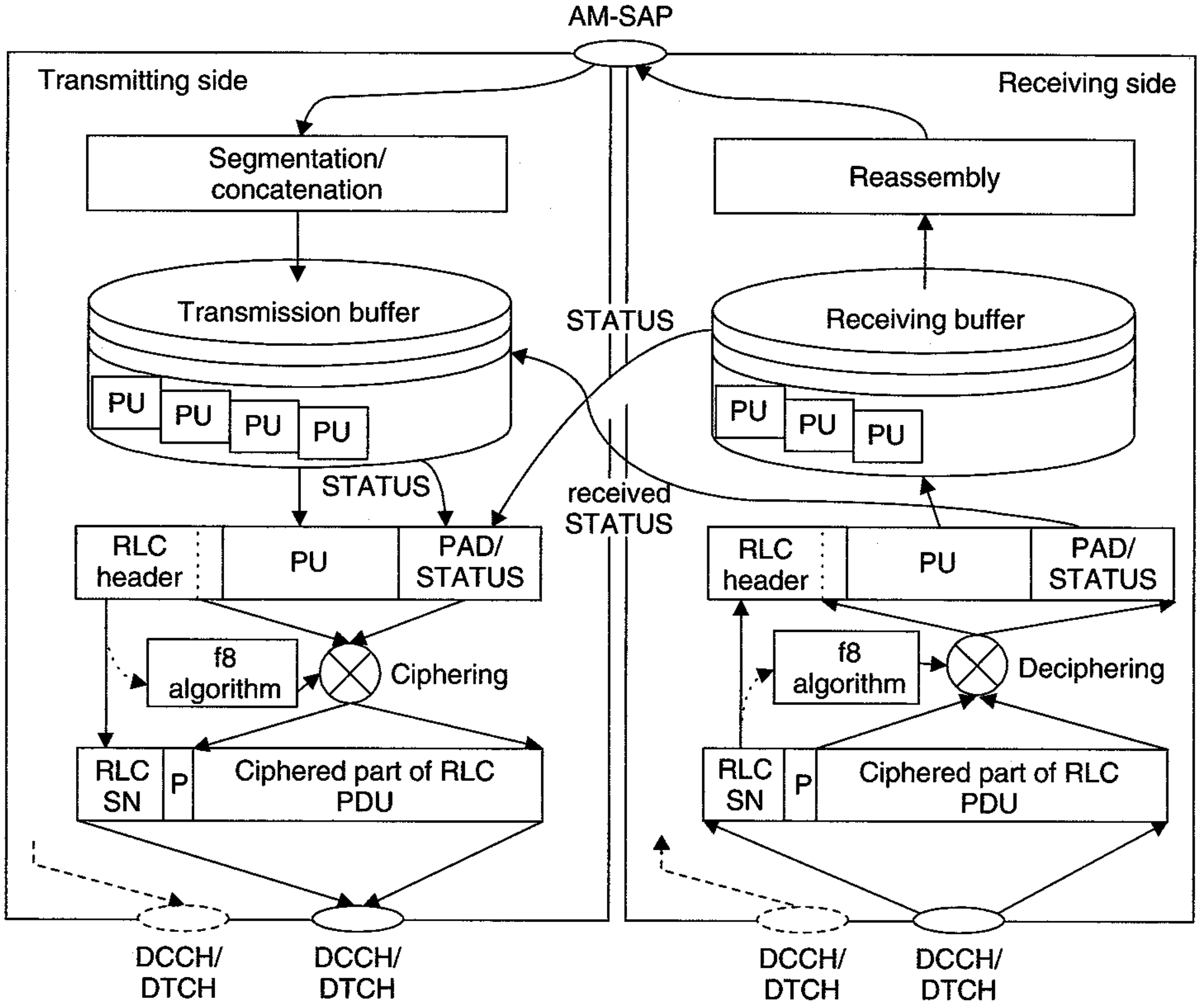


- Shows connection between logical channels and MAC-SAPs
- The CRC check result is delivered to the RLC together with data

RLC Functions

- Segmentation and Reassembly
- Concatenation
- Padding
- Transfer of user data
- Error Correction
- Sequential delivery of packets
- Duplicate Detection
- Flow Control
- Sequence Number Check
- Protocol error detection and recovery
- Suspend/Resume function for data transfer

RLC Example


AM-SAP
Transmitting side
Receiving side
Segmentation/
concatenation
Reassembly
Transmission buffer
Receiving buffer
STATUS
PU
PU
PU
PU
PU
PU
PU
STATUS
received
STATUS
RLC
header
PU
PAD/
STATUS
RLC
header
PU
PAD/
STATUS
f8
algorithm
Ciphering
f8
algorithm
Deciphering
RLC
SN
P
Ciphered part of RLC
PDU
RLC
SN
P
Ciphered part of RLC
PDU
DCCH/
DTCH
DCCH/
DTCH
DCCH/
DTCH
DCCH/
DTCH

Recap

Recap

Recap

Recap

- Radio Resource Management
- Radio Resource Control Protocol
- Medium Access Control
- Radio Link Control Protocol

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Overview of Wireless Communications : Introduction to multi-carrier communications

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March 2004

Overview

Next two days...?

- Multi-carrier communications
- Wireless standardization activities

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Outline - Multi-carrier communications

- Introduction
- Multi-carrier transmission techniques
- Structure of OFDM modulation
- Power spectrum
- Performance in AWGN
- Performance in Fading
- Discrete representation and implementation
- Performance in Multipath fading

Introduction

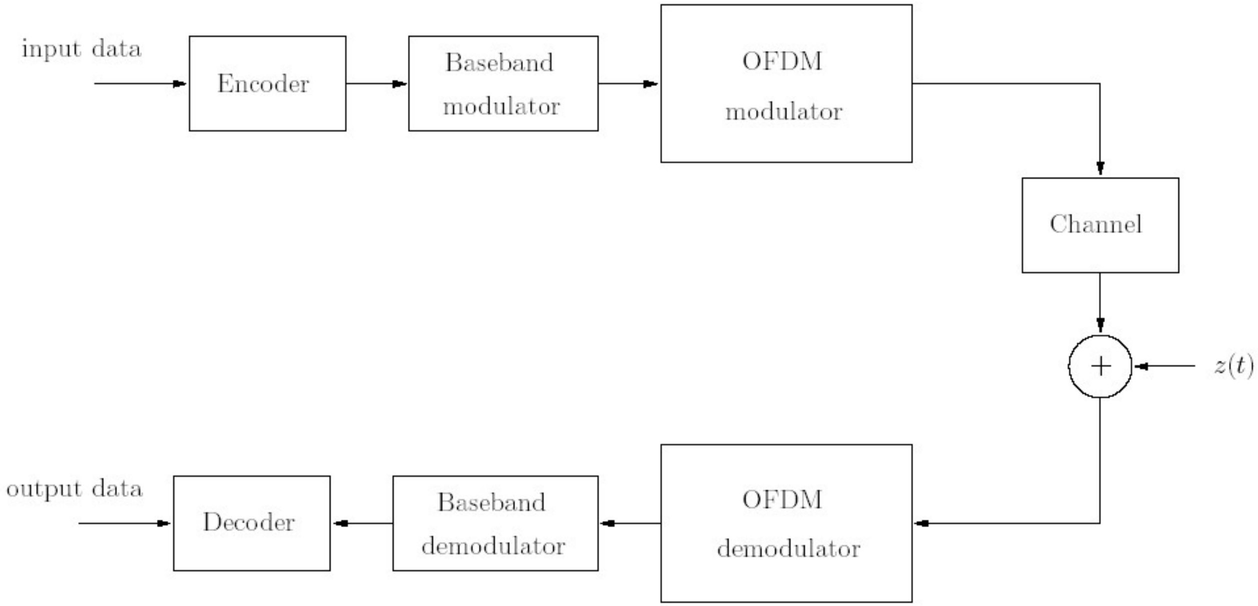
- Fading multipath channels introduce ISI into signals
- Increasing the transmission rate makes the time equalizer at the receiver more complex
  - Computational complexity
  - Delay due to processing
- This puts the limits on maximum achievable data rate
- OFDM is a possible solution

OFDM

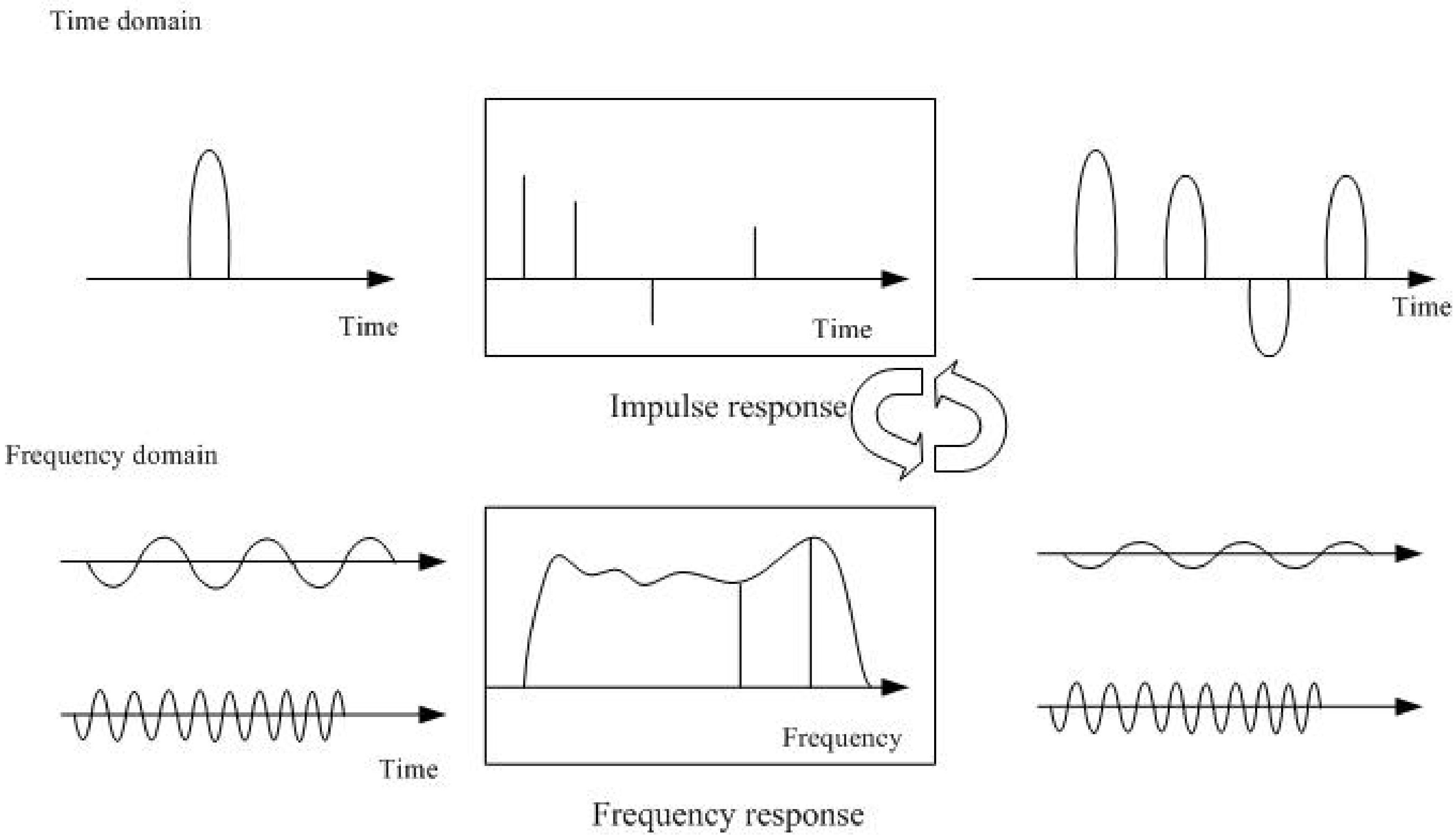
- OFDM was discovered in 60s but became popular only during the 90s (implementation reasons)
- OFDM - data through multiple frequency bands.
- Advantages
  - resistance to multipath propagation
  - Easy implementation with DSP technologies.
- Applications
  - Digital audio/video broadcasting (DAB/DVB)
  - broadband wireless loop
  - wireless ATM
  - IEEE 802.11a Multimedia Mobile Access Communication (MMAC)
  - HIPELAN/2
  - IEEE 802.16 BWA (Wireless MAN)

OFDM

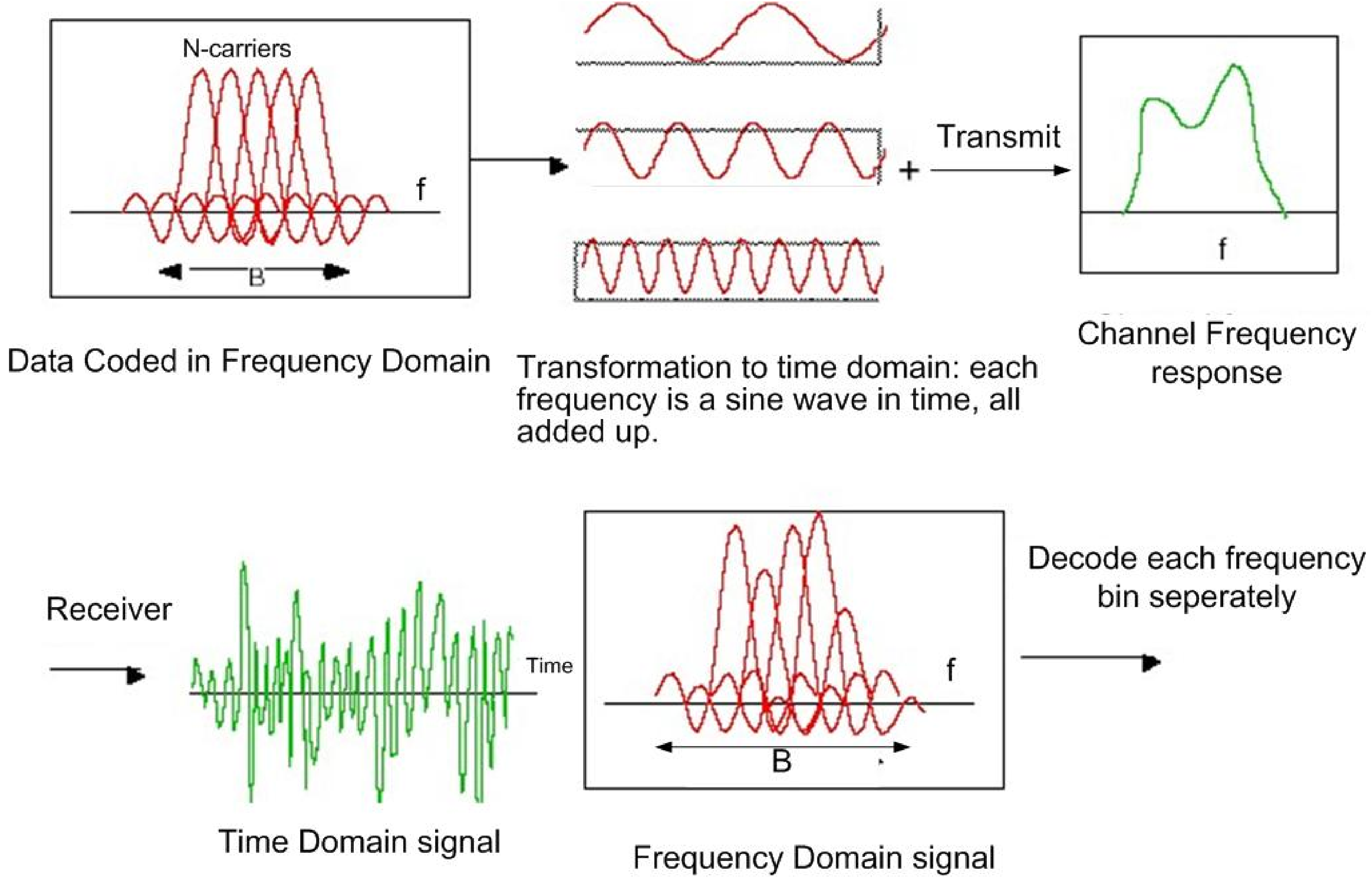
- OFDM is a modulation applied to the data-modulated signal
- OFDM splits the total bandwidth into several narrowbands
- Transform a high rate stream into several low-rate streams
- each low-rate stream is transmitted over a different subcarrier



Multipath channels



OFDM


N-carriers
f
B
Transmit
+
f
Data Coded in Frequency Domain
Transformation to time domain: each frequency is a sine wave in time, all added up.
Channel Frequency response
Receiver
Time
f
B
Decode each frequency bin seperately
Time Domain signal
Frequency Domain signal

Mathematical representation of OFDM signals

Orthogonality: In OFDM, subcarriers are overlapped but the signal can still be recovered without adjacent subcarrier interference because of the orthogonality between subcarriers.

Set of subcarriers $f_n(t), n = 0, 1, \ldots, N-1, t_1 \leq t \leq t_2$ is orthogonal if

$$\int_{t_1}^{t_2} f_n(t) f_m^*(t) dt = \begin{cases} 0, & n \neq m \\ \mathrm{K}, & n = m \end{cases}$$

where K is a constant independent of t, n or m. In OFDM the set of transmitted subcarriers can be written as

$$f_n(t) = \exp(j 2\pi f_n t)$$

where $j = \sqrt{-1}$ and the initial frequency offset

$$\begin{aligned} f_n &= f_0 + n\Delta f \\ &= f_0 + n/T. \end{aligned}$$

If the subcarriers are orthogonal We find

$$
\begin{aligned}
\int_{t_1}^{t_2} f_n(t) f_m^*(t) dt &= \int_{t_1}^{t_2} \exp(j2\pi(n-m)t/T) dt \\
&= \frac{\exp(j2\pi(n-m)t_2/T) - \exp(j2\pi(n-m)t_1/T)}{j2\pi(n-m)/T} \\
&= \frac{\exp(j2\pi(n-m)t_2/T)\left[1 - \exp(j2\pi(n-m)(t_1-t_2)/T)\right]}{j2\pi(n-m)/T} \\
&= 0 \quad \text{for} \quad (n \neq m) \;\; \& \;\; (n-m)(t_1-t_2)/T \quad \text{is an integer.}
\end{aligned}
$$

If the subcarriers are separated by $1/T$, they are orthogonal to each other provided $t_2 - t_1$ is an integer multiple of T

Baseband representation of OFDM signals

- An OFDM symbol consists of N subcarriers spaced by the frequency distance Δf.

- The total system bandwidth B is divided into N equidistant subchannels.

- All subcarriers are mutually orthogonal within a time interval of length $T = 1/\Delta f$.

- An OFDM symbol carries N-modulated data symbols $\mathbf{X}_m = (X_{m,0}, X_{m,1}, \ldots, X_{m,N-1})$

- Each $X_{m,n}$ carries $q = \log_2 M$ data bits (Normal values are $M = 2, 4, 8$ and so on), where m is a time index and n is a subcarrier index

- Each OFDM symbol thus carries $N \log_2 q$ data

- m-th OFDM block during the time interval T is formed

$$x_m(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_{m,n} \exp(j2\pi n \Delta f t) g_n(t - mT)$$

where $g_n(t)$ is a rectangular pulse applied to each subcarrier.

- The total continuous time signal $x(t)$ consisting of all OFDM blocks is

$$x(t) = \frac{1}{\sqrt{N}} \sum_{m=0}^{\infty} \sum_{n=0}^{N-1} X_{m,n} \exp(j2\pi n\Delta f t) g_n(t - mT)$$

- As different OFDM symbols do not overlap, we can consider a single OFDM symbol ($m = 0$) without loss of generality: $X_{m,n}$ may be replaced by X_n. Now we can write the OFDM signal as

$$x(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n e^{j2\pi\Delta f t} \quad 0 \leq t \leq T$$

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Discrete time representation of the OFDM symbol

- The discrete representation can be obtained by appropriate sampling

- As the bandwidth of an OFDM signal is $B = N\Delta f$, the signal can be completely determined by its samples if the sampling time $\Delta t = \frac{1}{B} = \frac{1}{N\Delta f}$.

$$x_k = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n e^{j2\pi kn/N}, \quad k = 0, 1, \ldots, N-1$$

- This equation describes exactly the N point inverse discrete Fourier transform (IDFT) of the input data $X_n, n = 0, 1, \ldots, N-1$.

- This can be easily implemented using inverse fast Fourier transform (IFFT) algorithms. The input symbols X_n represent digitally modulated binary data.

Bandpass representation of OFDM signals

- The baseband OFDM signal is modulated into a high frequency carrier f_c before the transmission

$$
\begin{aligned}
x_{bp}(t) &= \Re\{x(t)e^{j2\pi f_c t}\} \\
&= x_i(t)\cos(2\pi f_c t) - x_q(t)\sin(2\pi f_c t)
\end{aligned}
$$

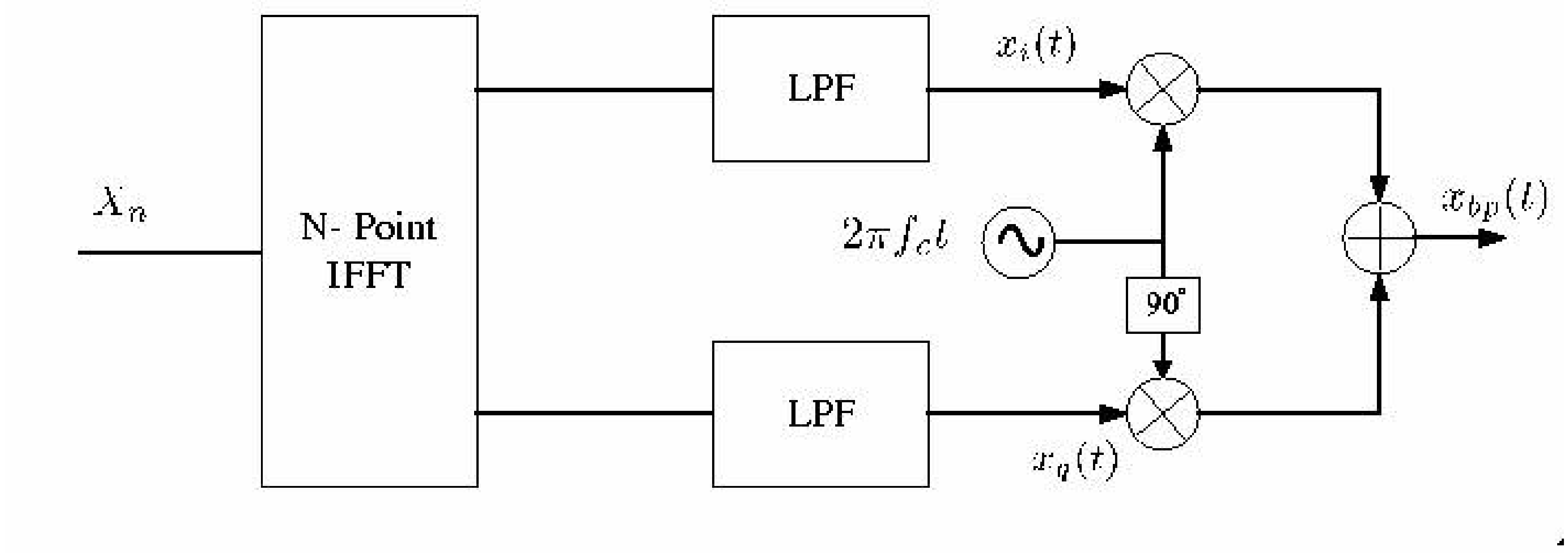
where $x(t)$ is the complex envelope or the low pass equivalent of the bandpass signal.

- This signal can expressed as

$$
x(t) = x_i(t) + jx_q(t)
$$

where $x_i(t)$ and $x_q(t)$ are the phase quadrature components of the OFDM symbol and are the real and imaginary outputs of the IDFT process.

bandpass signal generation at the transmitter



Bit Error Probability in AWGN

- The received sample at subcarrier k can be written as

$$r(t) = x(t) + z(t)$$

  – $x(t)$ - signal corresponding to the signal point of the used modulation
  – Complex gaussian random variable with zero mean and variance N_0

- The performance of OFDM over AWGN channels is the same as that of single carrier scheme that uses the same modulation

  – If BPSK modulation is used

$$P_b = \frac{1}{2}\text{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$$

  – The bandwidth efficiency in this case is given by

$$\eta = 1bits/s/Hz$$

OFDM signal over fading multipath channels

- In a fading multipath environment with coherence bandwidth B_m, the equivalent lowpass of the received signal is

$$r(t) = \sum_{i=0}^{L} h_i(t)x[t - \tau_i(t)] + z(t)$$

- With proper selection of N, we can make the OFDM block duration much longer than the maximum delay of the channel

$$T = NT_s \geq \tau = \frac{1}{2\pi B_c}$$

- The output sample of subcarrier n is given by

$$R_{m,n} = \frac{1}{\sqrt{T}} \int_{mT}^{(m+1)T} r(t) e^{-j2\pi \frac{n}{T} t} dt = H_n^{'} X_{m,n} + ISI + ICI + Z(n)$$

$$ISI = \sum_{l=0}^{N-1} G_{l,n} X_{m-1,l}$$

$$ICI = -\sum_{l=0, l \neq n}^{N-1} G_{l,n} X_{m,l}$$

OFDM needs to overcome the ISI and ICI .. how?

Solving the problem of fading multipath-guard interval in OFDM

- To solve this problem, every OFDM block is extended by a guard interval T_G

$$x(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_{m,n} e^{-j2\pi \frac{k}{T} t}, \;\; mT - T_G \leq t < (m+1)T$$

where $T_G \geq \tau$

- The guard interval is ignored at the receiver

$$r(t) = \sum_{n=0}^{N-1} H_n X_{m,n} e^{j2\pi n fract T}, \;\; mT \leq t < (m+1)T$$

where H_n is the transfer function of the channel $H_n = \sum_{l=0}^{L-1} h_l e^{j2\pi l n/N}$

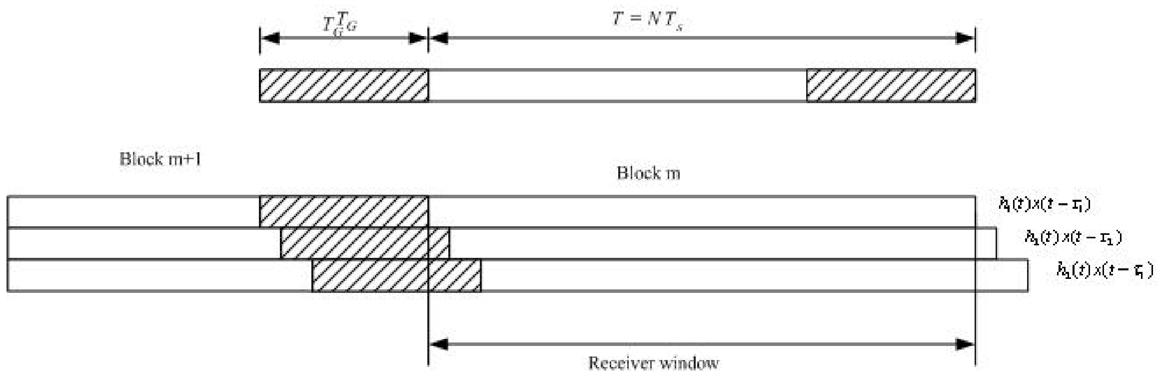
Guard interval in OFDM

- The output sample of the subcarrier n becomes

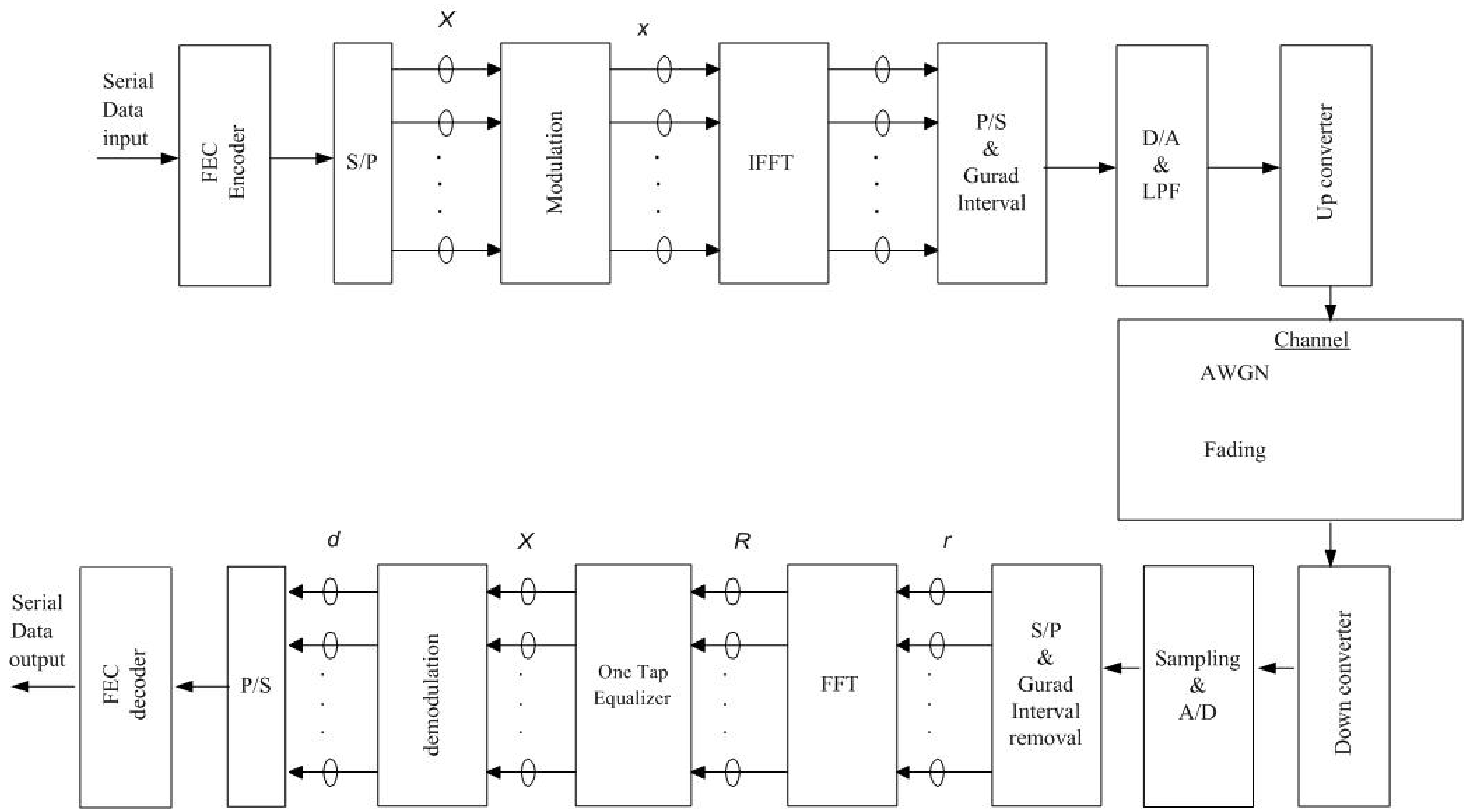
$$r_{m,n} = \int_{nT}^{(n+1)T} r(t) e^{-j2\pi \frac{n}{T} t} dt = H_n X_{m,n} + Z_{m,n}$$

  which is ISI free

- The OFDM with guard interval has transformed a frequency selective fading into N parallel flat fading channels



Component of an OFDM system



Performance in OFDM signals in Frequency selective fading channels

- Received signal

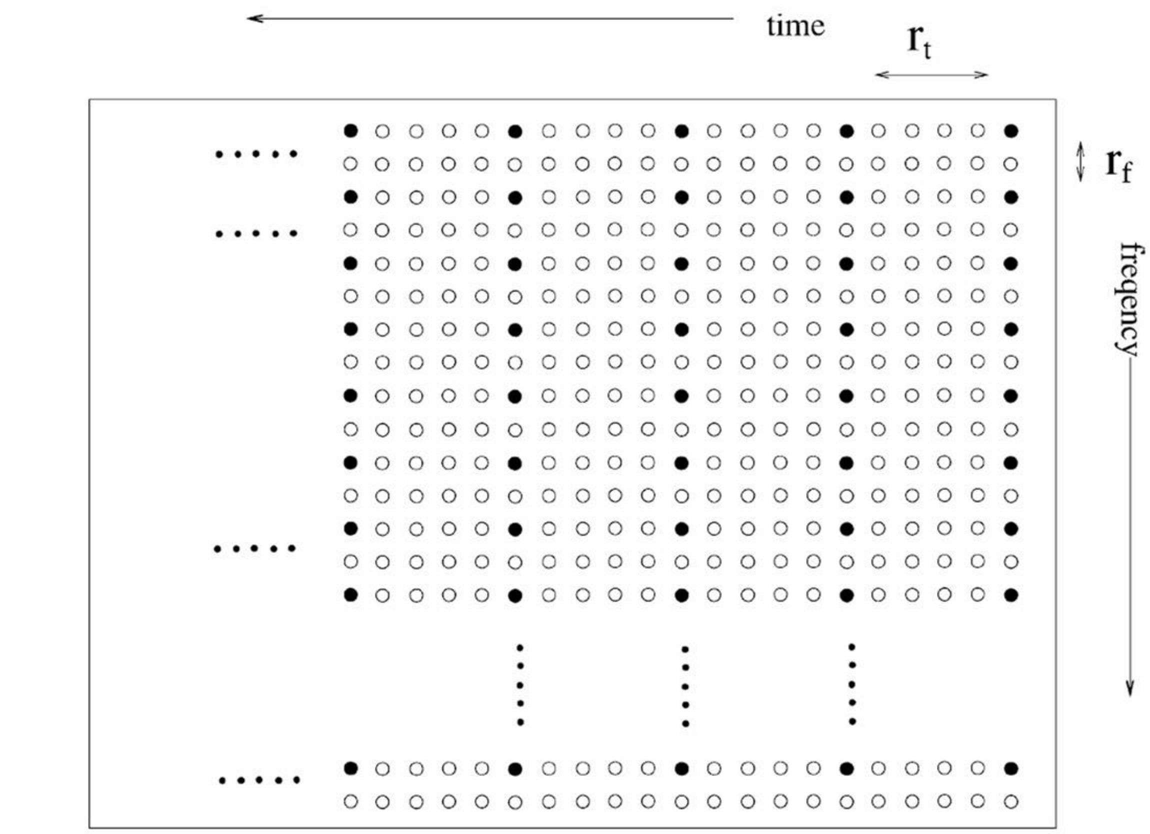
$$R_{m,n} = H_n S_{m,n} + Z_n$$

- This is again same as single carrier performance
  - For BPSK

$$P_b = \frac{1}{2}\left[1 - \sqrt{\frac{2\sigma^2 E_b/N_0}{1 + 2\sigma^2 E_b/N_o}}\right]$$

  - OFDM solves the problem of ISI but not the problem of signal fading

Typical pilot symbol distribution in time-frequency plane of an OFDM system



Dimensioning an OFDM system

The selection of suitable number of carriers for the OFDM system depends on several parameters.

$$(\mu N)_{\min} = \frac{R_b \tau_{max}}{\delta R_c}$$

where R_b is the data rate, τ_{max} is the maximum delay spread of the channel, δ is the proportion of the guard interval to the OFDM symbol duration, R_s is the code rate and N is the number of sub carriers.

Drawbacks

- High sensitivity to frequency errors
- Phase noise
- Sensitivity to timing errors
- High peak-to-average power ratios

Frequency offset is a problem in OFDM

- Transmitted signal with a frequency error

$$x(t) = \sum_{n=0}^{N-1} X_n e^{j2\pi(f_n+f_0)t}$$

- Received signal

$$r_k = \sum_{n=0}^{N-1} H_n X_n e^{j2\pi k(n+\epsilon)/N} + n_k$$

where $\epsilon = \frac{f_0}{\Delta f} = \frac{f_0}{1/(NT)}$

- Demultiplexed signal

$$R_n = \sum_{k=0}^{N-1} r_k e^{-j2\pi kn/N}$$

$$= X_n H_n \frac{\sin(\pi\epsilon)}{N\sin(\pi\epsilon/N)} e^{j\pi\epsilon(N-1)/N} + I_n + Z_n$$

where $I_n = \sum_{l=0}^{N-1} X_l H_l \frac{\sin(\pi\epsilon)}{N\sin(\pi(l-n+\epsilon/N))} e^{j\pi\epsilon(N-1)/N} e^{j\pi(l-n)/N}$

Phase noise

- Introduce by the local oscillator
- Can be considered as parasitic phase modulation
- Assuming only the phase noise $\phi_k, r_k = X_k e^{j\phi_k}$
- Assuming ϕ_k is small $e^{j\phi_k} \approx 1 + j\phi_k$
- The FFT output at the receiver

$$
\begin{aligned}
R_n &\approx X_n + \frac{j}{N} \sum_{i=0}^{N-1} X_i \sum_{k=0}^{N-1} \phi_k e^{j(2\phi/N)(i-n)k} \\
&\approx X_n + C_n
\end{aligned}
$$

Time offset effect

- Transmitted signal with a time offset τ

$$x'(t) = x(t-\tau)$$

- Received signal

$$R(t) = \int_0^T x(t-\tau)e^{-j2\pi n\Delta f t}dt$$

$$R_n X_n e^{-j2\pi n\tau/T}$$

- In the presence of a timing offset each data symbol is rotated by a phase angle given by

$$\Phi = 2\phi n\tau/T$$

Peak-to-average power ratio problem

- OFDM suffers from high PAR defined as,

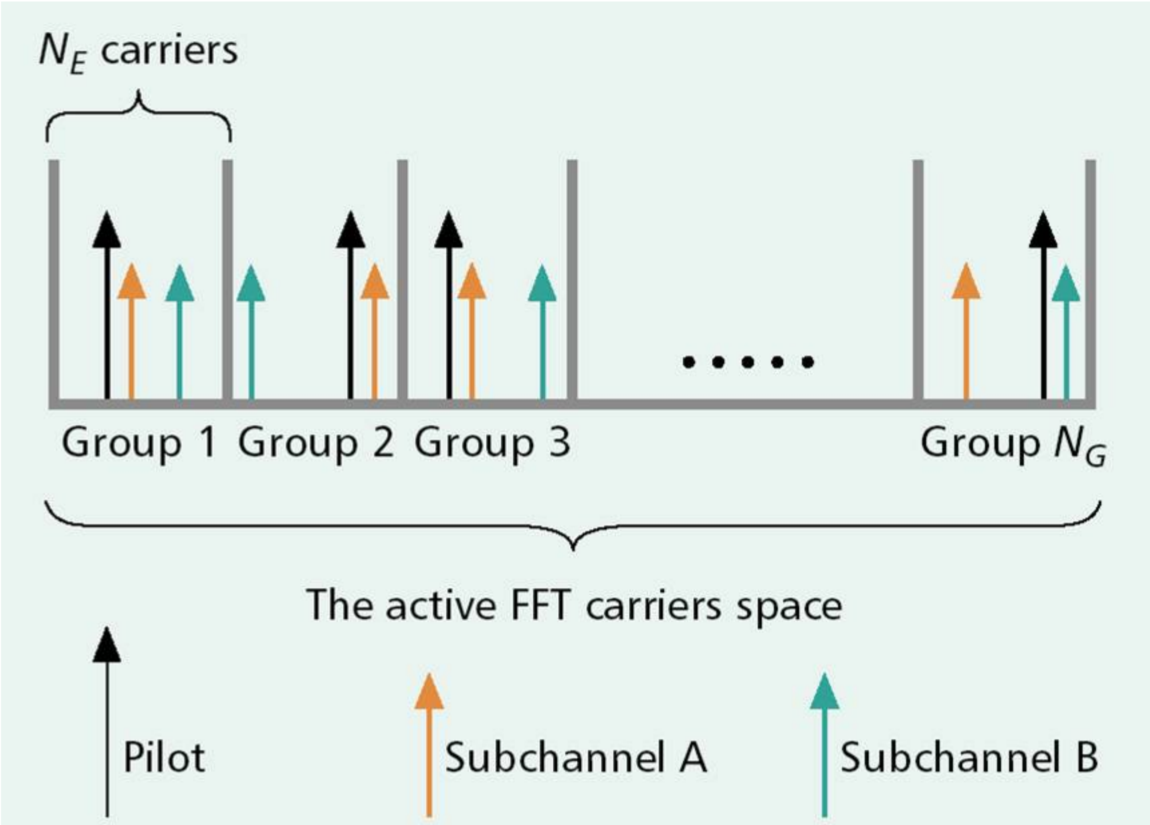
$$\xi = \frac{\max_{t\in[0,T)} |x(t)|^2}{P_{av}} = \frac{\max_{t\in[0,T)} |x(t)|^2}{E\{|x(t)|^2\}}$$

- Maximum PAR of an N subcarrier OFDM signal

$$\begin{aligned}
\xi &= \max_{t\in[0,T)} |x(t)|^2 \\
&= \max_{t\in[0,T)} \left| \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n e^{j2\pi\Delta f t} \right|^2 \\
&\leq \frac{1}{N} \left[\sum_{n=0}^{N-1} \left| X_n e^{j2\pi\Delta f t} \right| \right]^2 \\
&\leq N.
\end{aligned}$$

Muticarier-Multiple access schemes..!

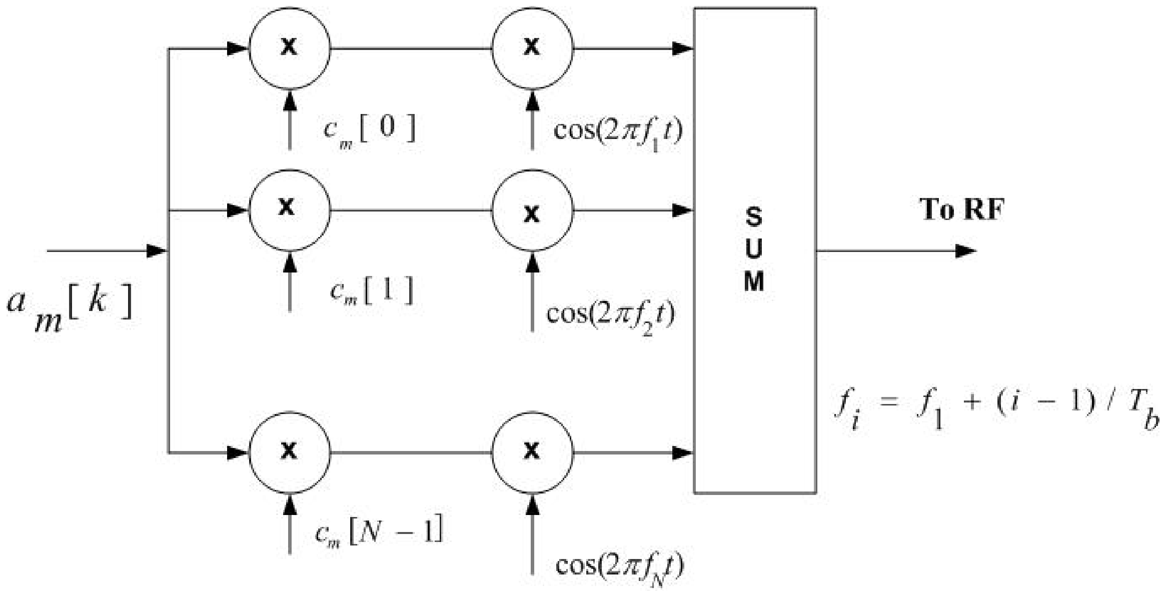
Orthogonal Frequency division multiple access (OFDMA)



BWA - IEEE 802.16a

Muticarier-Multiple access schemes..!

Muticarier-CDMA



3G - UMTS

Other - OFDM/TDMA

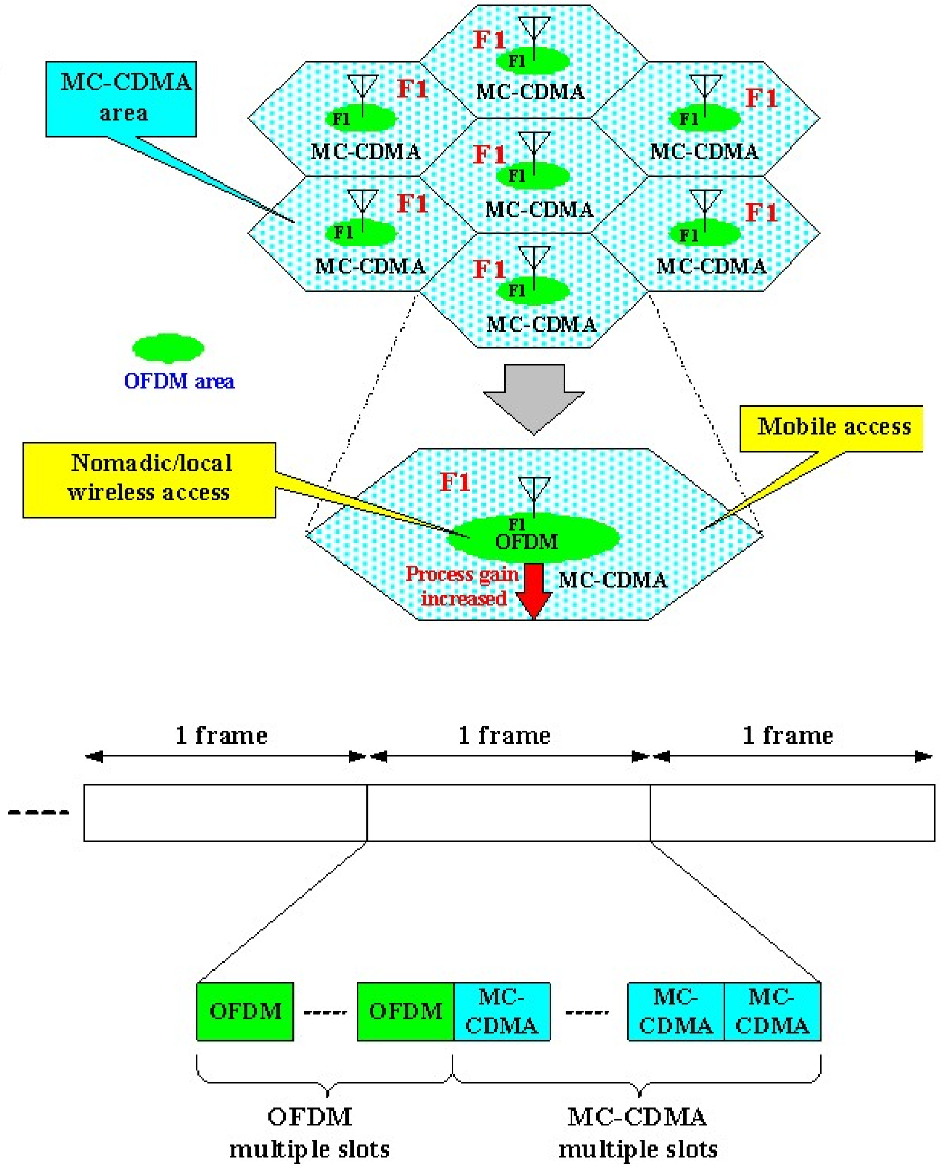
Muticarier-Multiple access schemes

- The link quality of the OFDM system is degraded when the co-channel interference signal level from adjacent cells is increased

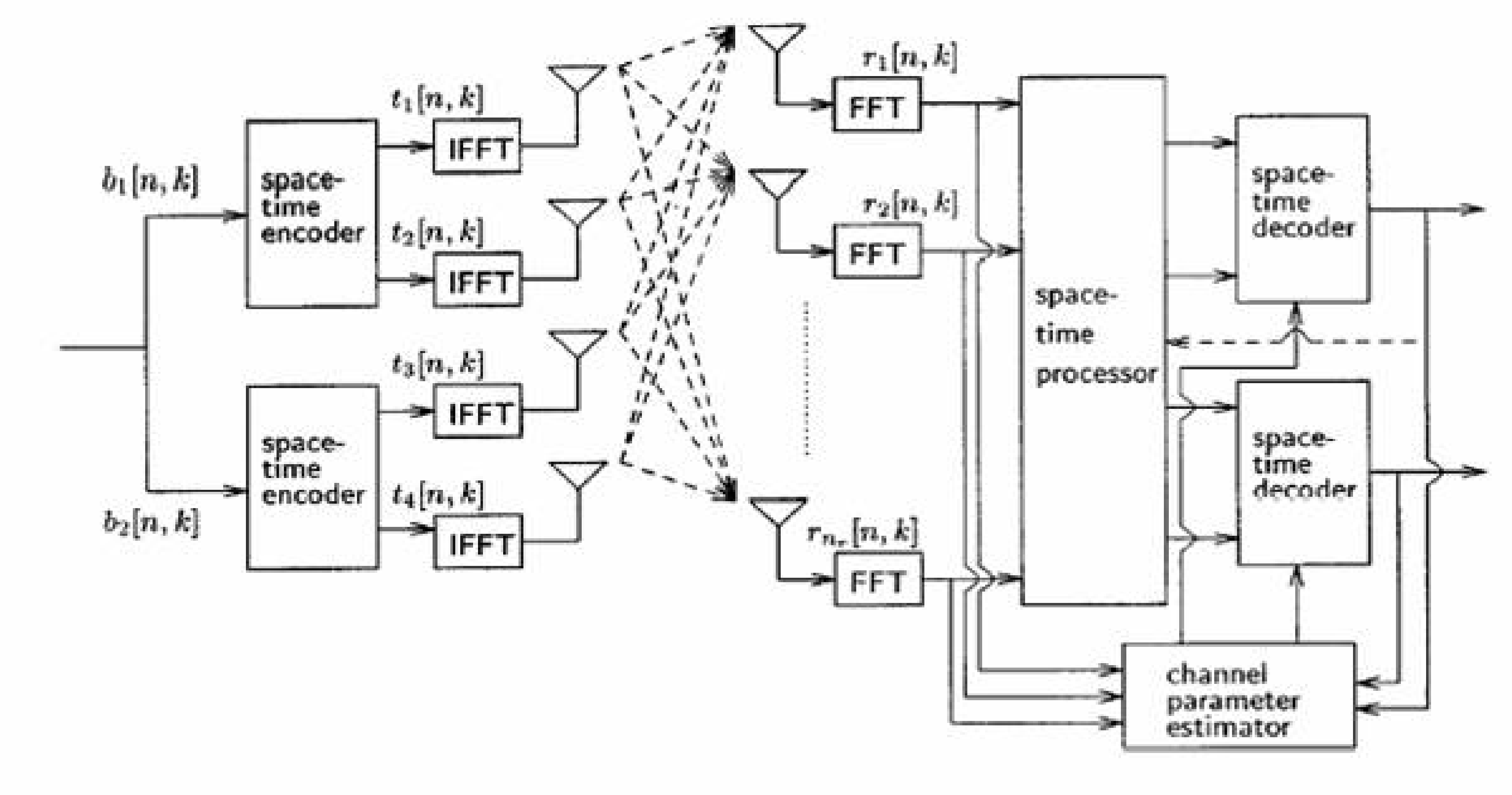
- The spread spectrum technique has tolerance to co-channel interference but it is difficult to enhance its transmission rate per user by restriction of allocated bandwidth

- Solution..! Combined approach.. OFDM/MC-CDMA

Example configuration From ITU technology trends



MIMO-OFDM



$b_1[n,k]$
$b_2[n,k]$
space-time encoder
space-time encoder
$t_1[n,k]$
$t_2[n,k]$
$t_3[n,k]$
$t_4[n,k]$
IFFT
IFFT
IFFT
IFFT
$r_1[n,k]$
$r_2[n,k]$
$r_{n_r}[n,k]$
FFT
FFT
FFT
space-time processor
space-time decoder
space-time decoder
channel parameter estimator

OFDM applications

System	Data rates	Technology	Range	Mobility	Frequency range	Original application area
GSM (including GPRS, HSCSD and EDGE)	9.6 kb/s up to 384 kb/s	TDMA, FDD	Up to 35 Km in GSM, lower for data	High	900, 1800, 1900 MHz	Public and private environment
IMT-2000, UMTS (UTRA)	Max. 2 Mb/s	IMT-2000 family, WCDMA (FDD) + TD-CDMA (TDD)	30 m–20 Km	High	2 GHz (ITU spectrum)	Public and private environment
DECT/DECTlink	Max. 2 Mb/s	TDMA/TDD	Up to 50 m	Low	1880–1900 MHz	Office and residential environment
Bluetooth	Max. 721 kb/s	Direct sequence or frequency hopping	0.1–10 m	Very low	2.4 GHz ISM band	Cable replacement, SoHo environment
HIPERLAN 2	25 Mb/s	OFDM, TDD	50–300 m	Low	5 GHz	Corporate environment, public hot spots
IEEE 802.11a	About 20 Mb/s	OFDM, TDD	50 - 300 m	Low	5 GHz	Corporate environment, public hot spots
HIPERACCESS	About 25 Mb/s	Not yet specified	2 … 10 km	No	5–40 GHz	Business access, feeder
DAB	1.5 Mb/s	OFDM	≤ 100 km	High	E.g., 176–230 MHz 1452–1467.5 MHz	Audio broadcasting
DVB-T	5–31 Mb/s per 8 MHz channel (mobile: 5–8, fixed 16–31)	OFDM	≤ 100 km	Medium to high	TV bands below 860 MHz	Video broadcasting
Cable modem	Down < 40 Mb/s Up < 10 Mb/s	FDD QAM/QPSK	5 to ~20 km	No	Down ~60 to 860 MHz Up 10 to ~40 MHz	Residential environment
ADSL	Down ≤ 6.144 (8) Mb/s Up (0.640 Mb/s)	DMT (carrierless AM/PM CAP)	2–6 Km	No	Baseband	SoHo (small office-home office), SME, residential environment

Conclusions

- OFDM is a modulation technique that splits a wideband signal into several narrow bands
- Higher bandwidth efficiency
- Frequency selective fading $\rightarrow$ Frequency flat fading
- No complex equalization

Overview of Wireless Communications : Wireless standardization activities

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March 2004

Why Standards

- Enables interoperability
- Encourages innovation
- Opens up new markets
- Creates trust and confidence in products
- Expands the market, brings down costs and increases competition
- Helps prevent the duplication of effort

Standardisation bodies

- International Telecommunications Union (ITU)
- Institute of Electrical and Electronics Engineers (IEEE)-SA
- The European Telecommunications Standards Institute (ETSI)
- American National Standards Institute (ANSI)
- Australian Communications Authority (ACA)

ITU

The ITU, headquartered in Geneva, Switzerland is an international organization within the United Nations System where governments and the private sector coordinate global telecom networks and services.

- Radio- communication (ITU-R)
- Telecom Standardization (ITU-T)

Recent standardization activities : International Mobile Telecommunications-2000 (IMT-2000) is the global standard for third generation (3G) wireless communications, defined by a set of interdependent ITU Recommendations. IMT-2000 provides a framework for worldwide wireless access by linking the diverse systems of terrestrial and/or satellite based networks. It will exploit the potential synergy between digital mobile telecommunications technologies and systems for fixed and mobile wireless access systems.

Radiocommunication Study Groups

- SG 1 - Spectrum management
- SG 3 - Radiowave propagation
- SG 4 - Fixed-satellite service
- SG 6 - Broadcasting services
- SG 7 - Science services
- SG 8 - Mobile, radiodetermination, amateur and related satellite services
- SG 9 - Fixed service
- CCV - Coordination Committee for Vocabulary
- CPM - Conference Preparatory Meeting
- SC - Special Committee on regulatory/procedural matters

Review of IEEE standardization process

- IEEE carries out standardisation through the IEEE standard association (IEEE-SA)
- IEEE-SA activities are credited by ANSI
- IEEE 802 LAN/MAN Standards Committee is one of the most important
- Ethernet (IEEE 802.3) is one of their standards

Active working groups

- **802.1** High Level Interface (HILI) Working Group
- **802.3** CSMA/CD Working Group
- **802.11** Wireless LAN (WLAN) Working Group
- **802.15** Wireless Personal Area Network (WPAN) Working Group
- **802.16** Broadband Wireless Access (BBWA) Working Group
- **802.17** Resilient Packet Ring (RPR)
- **802.18** Radio Regulatory Technical Advisory Group
- **802.19** Coexistence Technical Advisory Group
- **802.20** Mobile Wireless Access Working Group

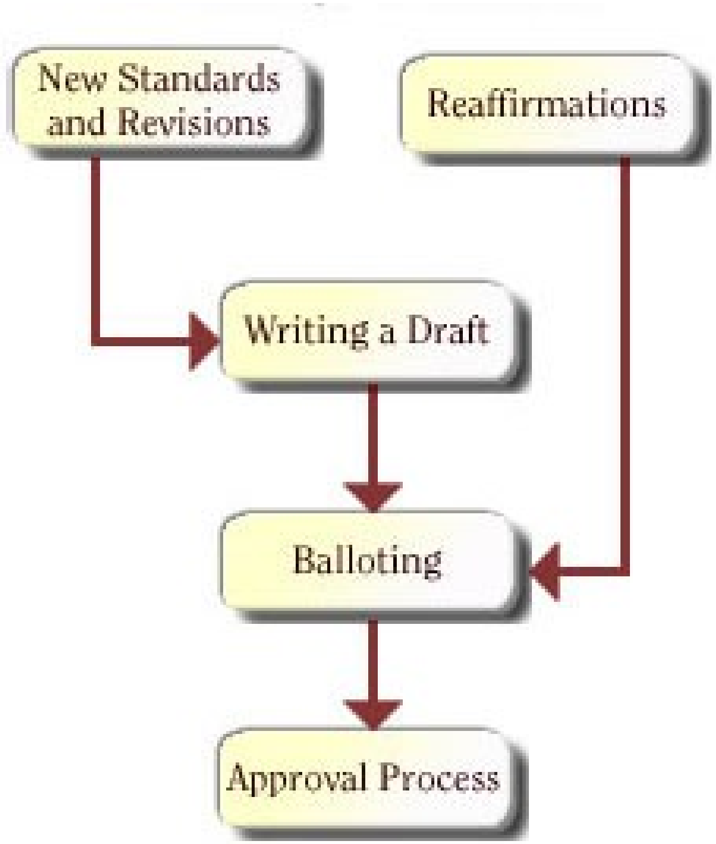
Standardization process

- Study group : Investigate the problem and consider the scope of and interest in a possible standardization project. PAR is made.

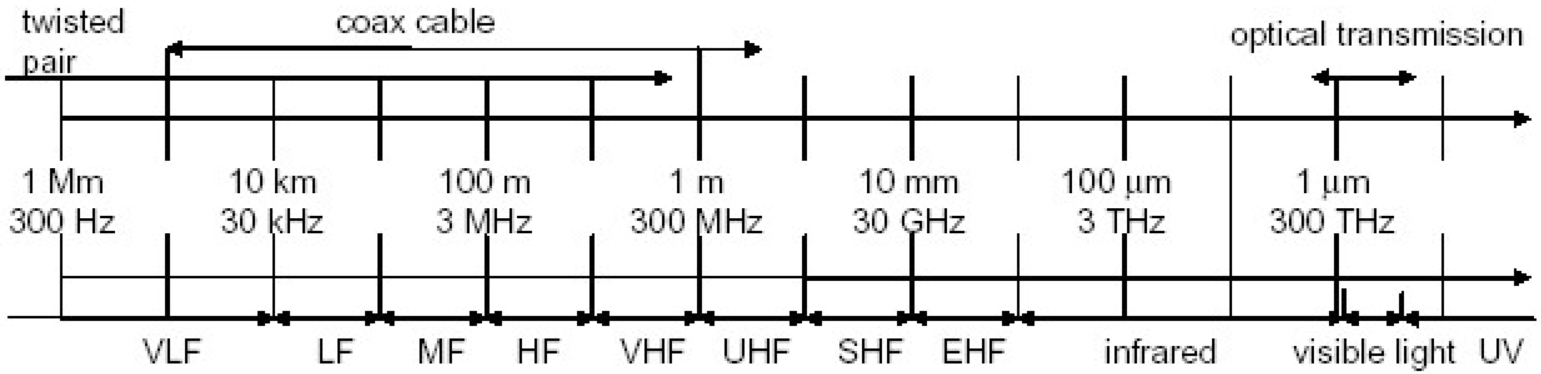
- Working group :Members -individuals (participation at meetings), 75% vote, draft

- Sponsor ballot : Wider group of interested individual, IEEE-SA members

Standards Process-at-a-Glance



Frequencies for wireless communications



VLF = Very Low Frequency
LF = Low Frequency
MF = Medium Frequency
HF = High Frequency
VHF = Very High Frequency

UHF = Ultra High Frequency
SHF = Super High Frequency
EHF = Extra High Frequency
UV = Ultraviolet Light

Frequency and wave length:

$\lambda = c/f$

Frequencies and Regulations

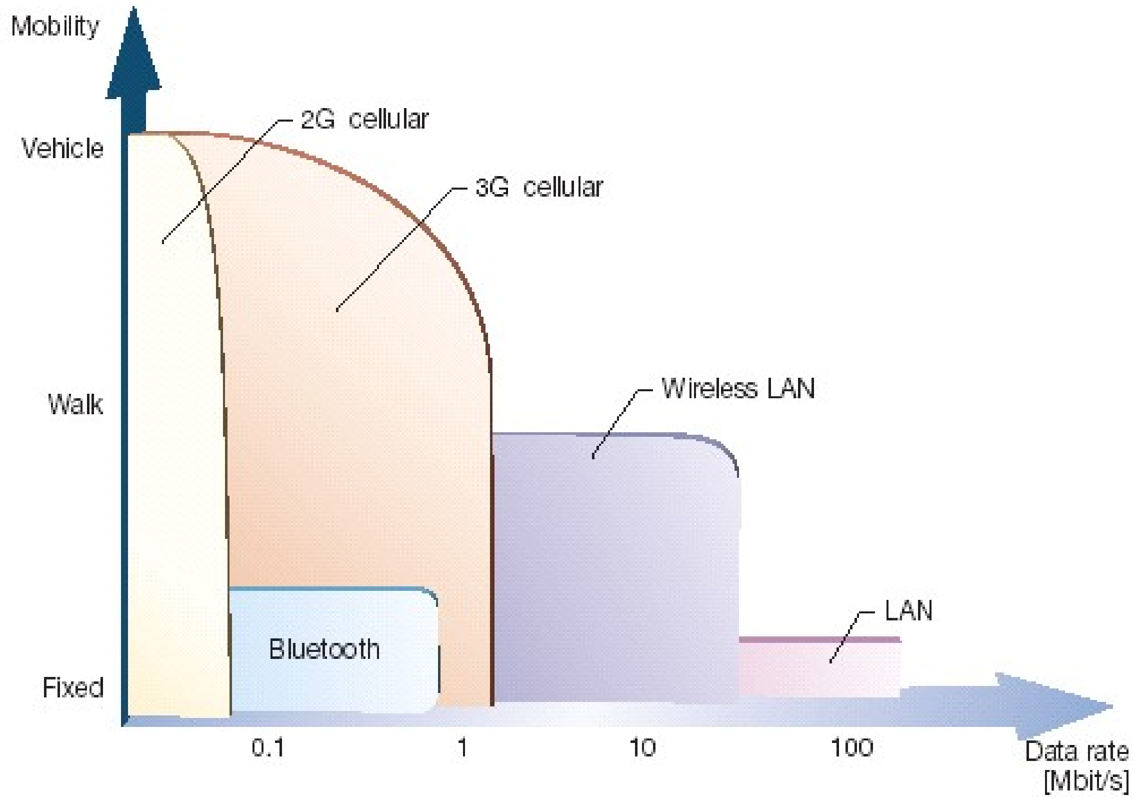
	Europe	USA	Japan
Cellular Phones	**GSM** 450-457, 479-486/460-467,489-496, 890-915/935-960, 1710-1785/1805-1880 **UMTS** (FDD) 1920-1980, 2110-2190 **UMTS** (TDD) 1900-1920, 2020-2025	**AMPS, TDMA, CDMA** 824-849, 869-894 **TDMA, CDMA, GSM** 1850-1910, 1930-1990	**PDC** 810-826, 940-956, 1429-1465, 1477-1513
Cordless Phones	**CT1+** 885-887, 930-932 **CT2** 864-868 **DECT** 1880-1900	**PACS** 1850-1910, 1930-1990 **PACS-UB** 1910-1930	**PHS** 1895-1918 **JCT** 254-380
Wireless LANs	**IEEE 802.11** 2400-2483 **HIPERLAN 2** 5150-5350, 5470-5725	902-928 **IEEE 802.11** 2400-2483 5150-5350, 5725-5825	**IEEE 802.11** 2471-2497 5150-5250
Others	**RF-Control** 27, 128, 418, 433, 868	**RF-Control** 315, 915	**RF-Control** 426, 868

ITU-R holds auctions for new frequencies, manages frequency bands worldwide.

Wireless Networks

- Wide area networks (WAN)
- Local area networks (LAN)
- Personal area networks (PAN)

Mobility vs Data Rate



Mobile Generations

1981 Analog cellular radio: 1^{st} Generations (AMPS, ETACS)

- reachability
- voice

1990 Digital cellular radio: 2^{nd} Generations

- High capacity
- Improved performance
- lower cost
- security
- voice, data

2002 Wideband cellular radio: 3^{rd} generation

- increased data rate
- packet services
- multimedia services

Mobile Generations

	1980s	1990s	2000s	2010s	2020s
Generation	First	Second	Third	Fourth	Fifth
Keywords	Analog	Digital personal	Global world standards	High data rate High mobility IP-based	High data rate High mobility IP-based
Systems	Analog cellular	Digital cellular	IMT-2000	4G-cellular	5G-cellular
		GSM, IS-54, PDC	(3G-cellular)	Broadband access	Broadband access
	Analog cordless	Digital cordless		ITS, HAPS	ITS, HAPS
		DECT, PHS	Max data rate	Mini data rate	Mini data rate
			2 Mb/s	2–20Mb/s?	20–100 Mb/s?
		Mobile satellite			
		Iridium, Inmarsat-M			

1G Generation

Table 11.1 AMPS and ETACS Radio Interface Specifications

Parameter	AMPS Specification	ETACS Specification
Multiple Access	FDMA	FDMA
Duplexing	FDD	FDD
Channel Bandwidth	30 kHz	25 kHz
Traffic Channel per RF Channel	1	1
Reverse Channel Frequency	824–849 MHz	890–915 MHz
Forward Channel Frequency	869–894 MHz	935–960 MHz
Voice Modulation	FM	FM
Peak Deviation: Voice Channels Control/Wideband Data	±12 kHz ±8 kHz	±10 kHz ±6.4 kHz
Channel Coding for Data Transmission	BCH(40,28) on FC BCH(48,36) on RC	BCH(40,28) on FC BCH(48,36) on RC
Data Rate on Control/Wideband Channel	10 kbps	8 kbps
Spectral Efficiency	0.33 bps/Hz	0.33 bps/Hz
Number of Channels	832	1000

Leading 2G technologies

Most popular 2G standards includes three TDMA standard and one CDMA standard

TDMA

- GSM - (Global system mobile)- 8 time slots in 200 KHz radio channel: Europe, Asia, Australia, South America and some part of US.
- IS-136 - (NADC - North American Digital Cellular)- 3 time slots in 30 KHz radio channel: North America, South America and Australia
- Pacific Digital Cellular - A Japanese TDMA, similar to IS-136

CDMA

- IS-95 - 64 users on each 1.25 MHz channel: North America, Korea, Japan, China, Australia

IS-136 (IS-95 REV.C)

Table 11.2 USDC Radio Interface Specifications Summary

Parameter	USDC IS-54 Specification
Multiple Access	TDMA/FDD
Modulation	$\pi/4$ DQPSK
Channel Bandwidth	30 kHz
Reverse Channel Frequency Band	824–849 MHz
Forward Channel Frequency Band	869–894 MHz
Forward and Reverse Channel Data Rate	48.6 kbps
Spectrum Efficiency	1.62 bps/Hz
Equalizer	Unspecified
Channel Coding	7 bit CRC and rate 1/2 convolutional coding of constraint length 6
Interleaving	2 slot interleaver
Users per Channel	3 (full-rate speech coder of 7.95 kbps/user) 6 (with half-rate speech coder of 3.975 kbps/user)

Global System for Mobiles (GSM)

- A second generation system.
- GSM was the world's first cellular system to specify digital modulation and network level architectures and services.
- As of 2001, there were 350 million GSM subscribes worldwide.
- The GSM standards are set by European Technical Standard Institute (ETSI): http://www.etsi.org

GSM 1800 (DCS1800)

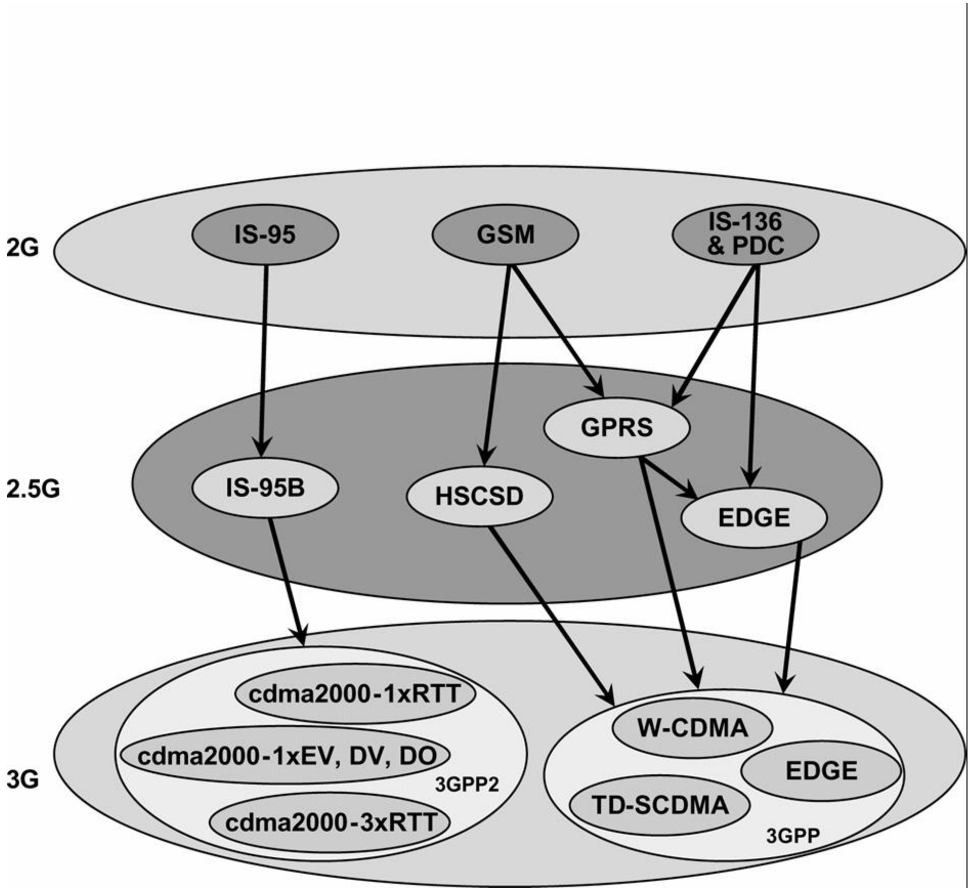
- An extension of the GSM system to the frequency range of 1710-1880 MHz (1710-1785 MHz Reverse, 1805-1880 MHz Forward)
- Low power MSs, Maximum power levels of MSs are 30 and 24dBm, which are significantly below the typical power levels of GSM 900
- Path loss is high. Needs more BSs to cover a given geographical area

GSM

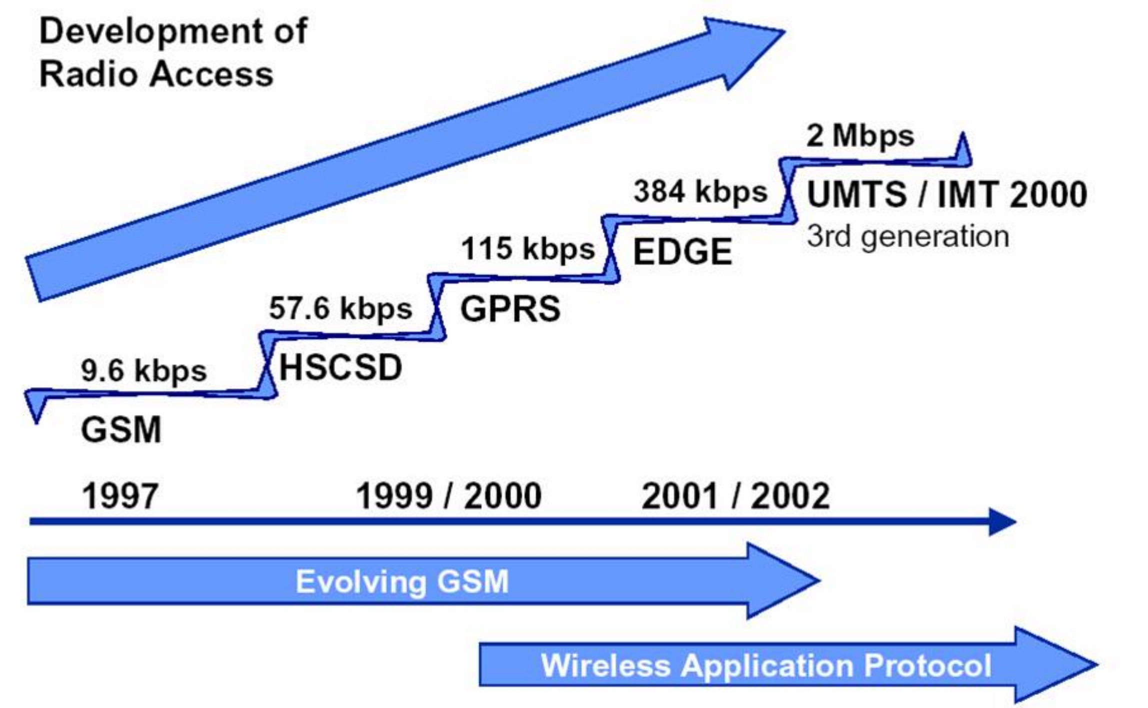
Table 11.3 GSM Air Interface Specifications Summary

Parameter	Specifications
Reverse Channel Frequency	890–915 MHz
Forward Channel Frequency	935–960 MHz
ARFCN Number	0 to 124 and 975 to 1023
Tx/Rx Frequency Spacing Tx/Rx Time Slot Spacing	45 MHz 3 Time slots
Modulation Data Rate	270.833333 kbps
Frame Period	4.615 ms
Users per Frame (Full Rate)	8
Time Slot Period	576.9 μs
Bit Period	3.692 μs
Modulation	0.3 GMSK
ARFCN Channel Spacing	200 kHz
Interleaving (max. delay)	40 ms
Voice Coder Bit Rate	13.4 kbps

Upgrade paths for 2G technologies



2G Evolution towards 3G



Upgrade paths for 2G technologies

HSCSD (High speed circuit switched data)

- Use up to 4 consecutive time slots in GSM, relax the FEC (14.4 kbps) - Maximum data rate of 57.6 kbps

GPRS - (General packet radio service)

- Packet oriented, suitable for non-real time data, all 8 time slots can be used - Maximum data rate 115 kbps

EDGE - (Enhanced data rates for GSM evolution)

- New modulation format 8-PSK, all 8 time slots can be used - Maximum data rate 384 kbs

If error protection is completely removed (21.4 kbps), data rates of GPRS and EDGE can be increased up to 171.2 kbps and 547.2 kbps respectively.

Enhanced data for GSM evolution (EDGE)

- New modulation format 8-PSK, all 8 time slots can be used - Maximum data rate 384 kbs
- the EDGE concept can provide capabilities of the 3rd generation systems within the frequency bands (800, 900, 1800 and 1900 MHz) of the existing 2nd generation systems
- Introduction of EDGE does not require addition of any new abstract machines in the structure of a GSM network that is already supporting GPRS
- However, the increased bit rates of EDGE impose heavy capacity requirements on interfaces of the GSM/GPRS network architecture
- Symbol rate 271 ksymbol/s is used for both modulations, leading to gross bit rates per time slot of 22.8 kbit/s and 69.2 kbit/s for GMSK and 8-PSK, respectively

EDGE

- When introducing the EDGE modulation scheme 8-PSK, higher SNR is required in a receiver for the same performance, due to the higher gross bit rate

- Most of the EDGE physical layer parameters are identical to those of GSM. The 8-PSK burst consists of:
  - 3 tail symbols, followed by
  - data block of 58 symbols, followed by
  - training sequence of 26 symbols, followed by
  - second data block of 58 symbols, followed by
  - 3 tail symbols, followed by
  - 8.25 guard symbols exactly like the normal burst of GSM

Wireless Application Protocol

- The Wireless Application Protocol (WAP) is a global standard for bringing Internet-type content and services to mobile phones and other wireless devices

- The WAP standards are not part of GSM standardization, but instead they are maintained by an industry consortium called the WAP Forum (www.wapforum.org)

- The WAP Forum was founded by mobile phone manufacturers - including Ericsson, Motorola and Nokia - in June 1997, and its membership now exceeds 500 organizations

- WAP standards do not specify, how data should be transmitted over the air interface of a wireless connection.

- WAP specifications are designed to operate over various bearer services of different digital cellular systems, including short message service, circuit-switched data, and packet data

CDMA digital cellular standard (IS95)

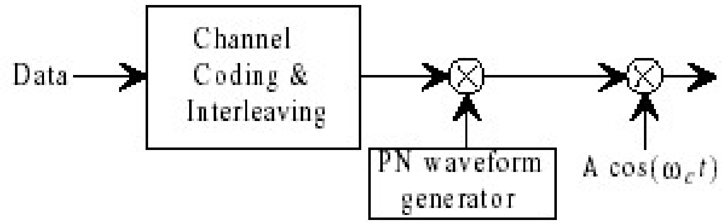
- The CDMA technology provides resistance against (multiple access) interference and multipath propagation, which make the use of CDMA attractive in mobile radio environment

- The IS-95 system is an asymmetrical system, where different modulation, spreading and channel coding techniques etc. are used in forward and reverse transmission

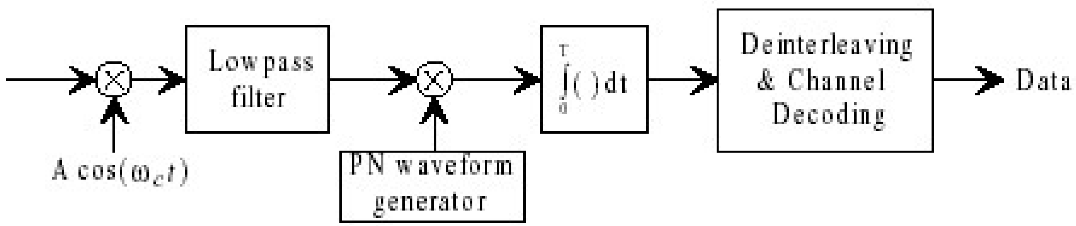
- The user data rate changes in real-time depending on the voice activity

- The speech coder is Qualcomm code exited linear predictive (QCELP) code

A typical CDMA system



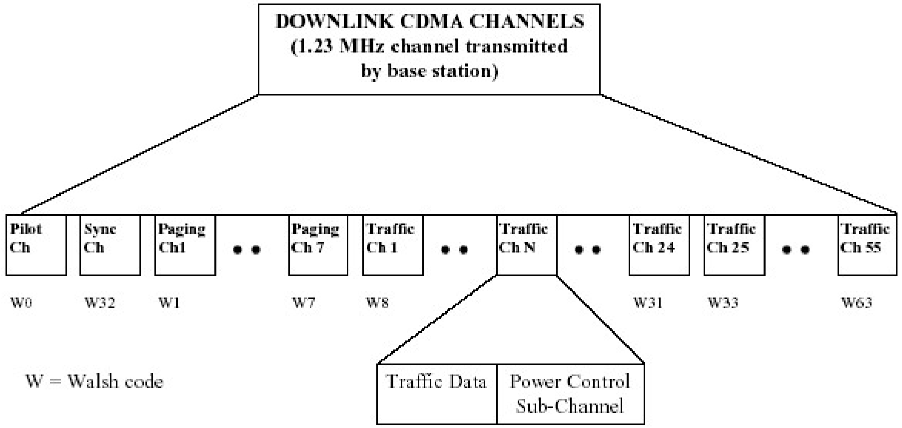
BPSK/CDMA Receiver:



IS-95 System parameters

	Uplink	Downlink
Frequency band [MHz]	824-849	869-894
Modulation	OQPSK	QPSK
Speech Coding	QCELP	QCELP
rate [kbit/s]	1.2/2.4/4.8/9.6	1.2/2.4/4.8/9.6
effective Tx rate [kbit/s]	1.2/2.4/4.8/9.6	9.6
Channel coding	rate 1/3 convl., CRC	rate 1/2 convl., CRC
coded rate [kbit/s]	28.8	19.2
Spreading code	pseudonoise	Walsh
Processing gain	4	64

Forward Channel



- Each individual channel is separated from other channels by its spreading code, which is one of the 64 Walsh codes W0 through W63.
- W0 is always used for pilot channel.
- W1 through W63 are used for traffic channels.

Forward Channel

Table 11.4 IS-95 Forward Traffic Channel Modulation Parameters Summary (does not reflect new 13.4 kbps coder)

Parameter	Data Rate (bps)			
User data rate	**9600**	**4800**	**2400**	**1200**
Coding Rate	1/2	1/2	1/2	1/2
User Data Repetition Period	1	2	4	8
Baseband Coded Data Rate	19,200	19,200	19,200	19,200
PN Chips/Coded Data Bit	64	64	64	64
PN Chip Rate (Mcps)	1.2288	1.2288	1.2288	1.2288
PN Chips/Bit	128	256	512	1024

Forward Channel

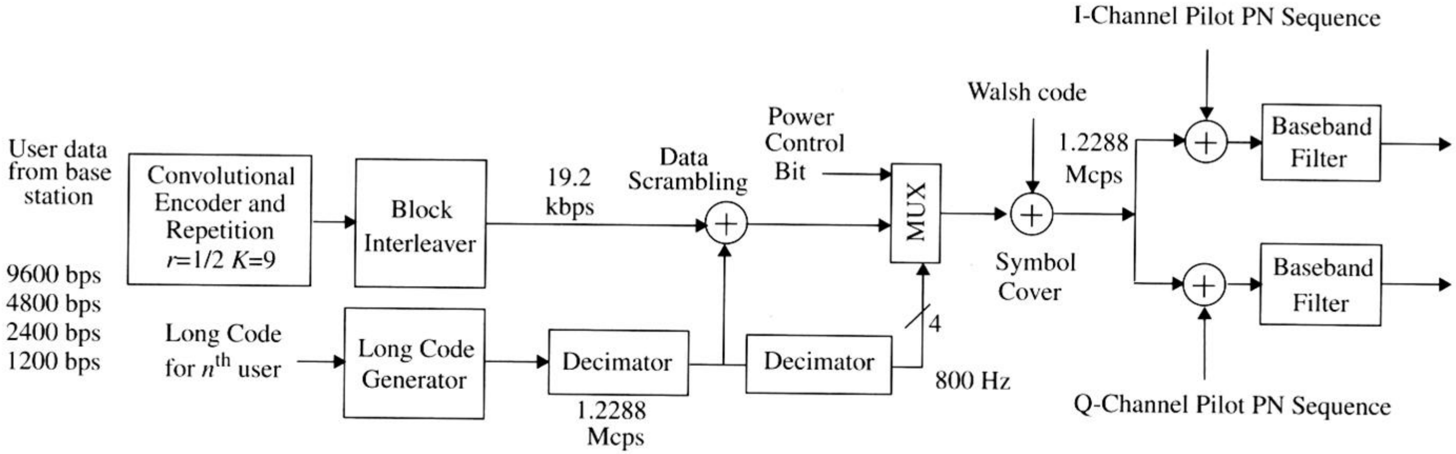


Figure 11.14 Forward CDMA channel modulation process.

Generation of Walsh-Hadamard codes

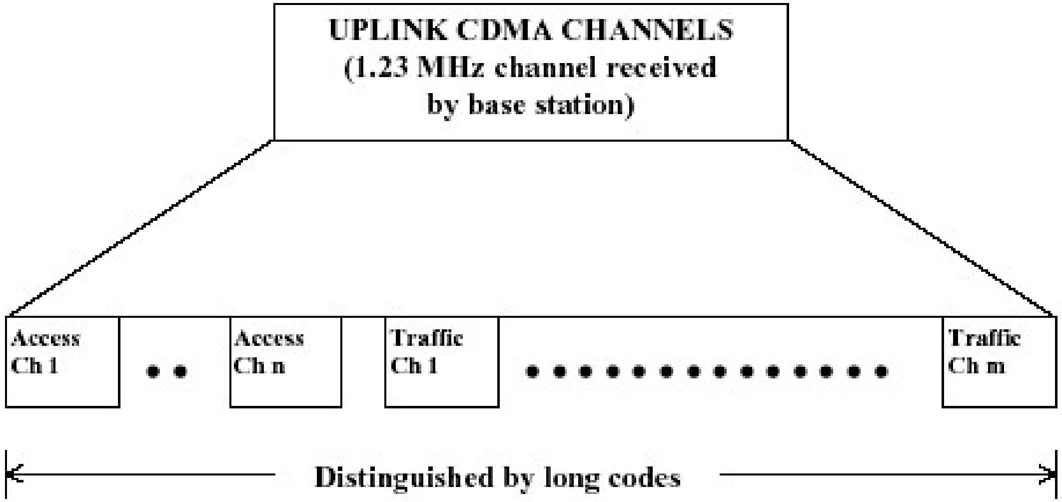
$$\mathbf{H}_1 = [0]$$

$$\mathbf{H}_{2m} = \begin{bmatrix} \mathbf{H}_m & \mathbf{H}_m \\ \mathbf{H}_m & \overline{\mathbf{H}}_m \end{bmatrix}, \quad \text{for } \mathrm{m} = 2^{\mathrm{J}}, \mathrm{J} \geq 1$$

Thus,

$$\mathbf{H}_2 = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}, \quad \mathbf{H}_4 = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{bmatrix}, \quad \text{etc.}$$

Reverse Channel



- Each uplink channel is separated from each other by the its specific long code. The period of the long code is 2^{42}-1 chips, and there are 2^{42}-1 different long codes, so that there are theoretically $2^{42} - 1$ distinct uplink channels.
- In one cell only up to 63 traffic channels may be used. One long code is permanently and uniquely associated with each mobile station
- Long code simplifies handover, since the code does not have to change on handover

Reverse Channel

Table 11.6 Reverse Traffic Channel Modulation Parameters Summary (does not reflect recent 13.4 kbps coder)

Parameter	Data Rate (bps)			
User data rate	**9600**	**4800**	**2400**	**1200**
Code Rate	1/3	1/3	1/3	1/3
TX Duty Cycle (%)	100.0	50.0	25.0	12.5
Coded Data Rate (sps)	28,800	28,800	28,800	28,800
Bits per Walsh Symbol	6	6	6	6
Walsh Symbol Rate	4800	4800	4800	4800
Walsh Chip Rate (kcps)	307.2	307.2	307.2	307.2
Walsh Symbol Duration (μs)	208.33	208.33	208.33	208.33
PN Chips/Code Symbol	42.67	42.67	42.67	42.67
PN Chips/Walsh Symbol	256	256	256	256
PN Chips/Walsh Chip	4	4	4	4
PN Chip Rate (Mcps)	1.2288	1.2288	1.2288	1.2288

Reverse Channel

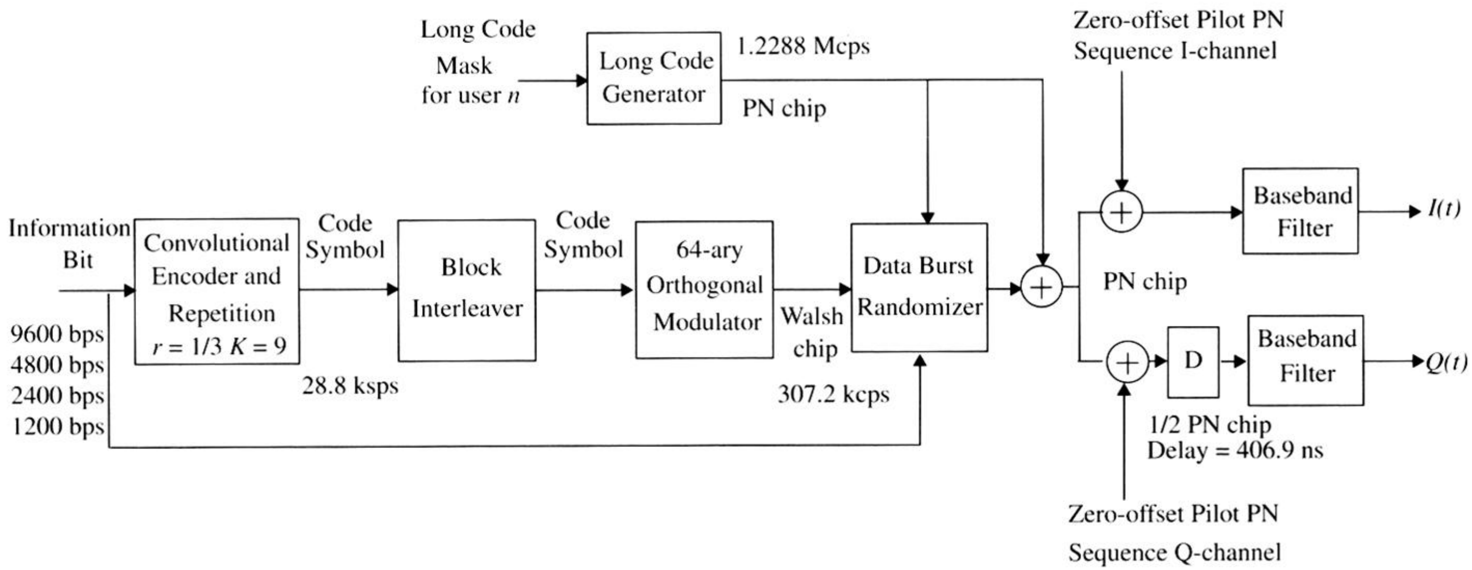


Figure 11.17 Reverse IS-95 channel modulation process for a single user.

Other important topics on IS-95A

- Power Control
- Access channel (Reverse), Sync channel (W32), pilot channel (W0), traffic channel, paging channel (W1-W7)
- Handoff
- Authentication
- Encryption

Expectations of 3G systems

- Voice quality comparable to the public switched telephone network
- 144 kbps data rate available to users in high-speed motor vehicles over large areas
- 384 kbps available to pedestrians standing or moving slowly over small areas
- Support for 2.048 Mbps for office use Symmetrical / asymmetrical data transmission rates
- Support for both packet switched and circuit switched data services
- More efficient use of the available spectrum in general
- Support for a wide variety of mobile equipment
- Flexibility to allow the introduction of new services and technologies, Voice Messaging email, fax, etc. Medium-rate multimedia Internet access, educational

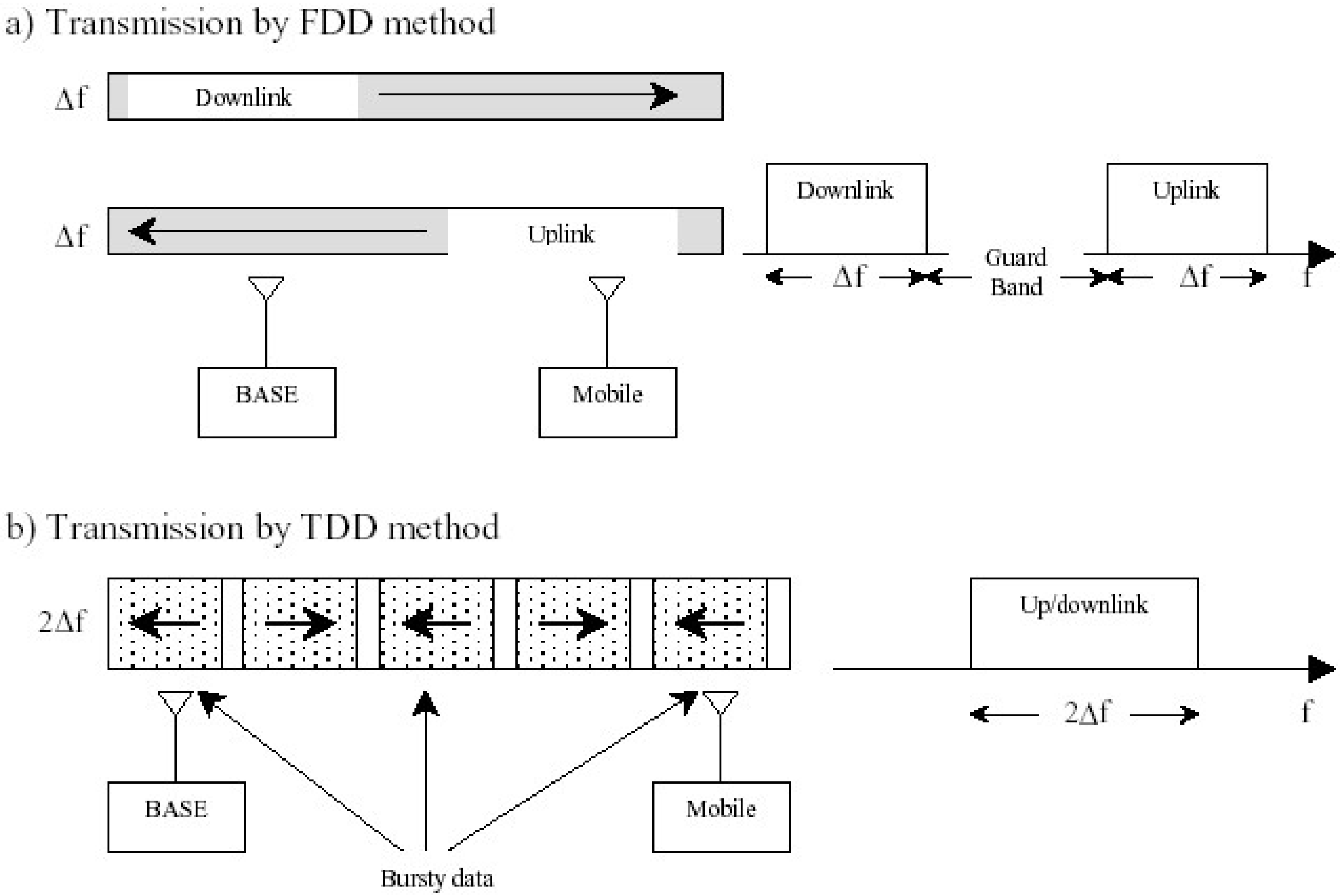
High-rate multimedia file transfer, video High-rate interactive multimedia video teleconferencing, telemedicine, etc.

- Mobility: quasi-stationary to high-speed platforms Global roaming: ubiquitous, seamless coverage Evolution from second generation systems

WCDMA vs cdma2000

Parameter	W-CDMA	cdma2000
Carrier spacing	5 MHz	3.75 MHz
Chip rate	3.84 Mcps	3.6864 Mcps
Data modulation	BPSK	FW – QPSK; RV - BPSK
Spreading	Complex (OQPSK)	Complex (OQPSK)
Power control frequency	1500 Hz	800 Hz
Variable data rate implement.	Variable SF; multicode	Repeti., puncturing, multicode
Frame duration	10 ms	20 ms
Coding	Turbo and convolutional	Turbo and convolutional
Base stations synchronized?	Asynchronous	Synchronous
Base station acquisition/detect	3 step; slot, frame, code	Time shifted PN correlation
Forward link pilot	TDM dedicated pilot	CDM common pilot
Antenna beam forming	TDM dedicated pilot	Auxiliary pilot

WCDMA duplexing modes



Channel Types in WCDMA

The following logical channels have been defined for WCDMA. The three available common control channels are:

- Broadcast control channel (BCCH) for system and cell specific information
- Paging channel (PCH) for messages to the mobiles in the paging area
- Common control channel (CCCH) for messages from network to a MS in one cell.

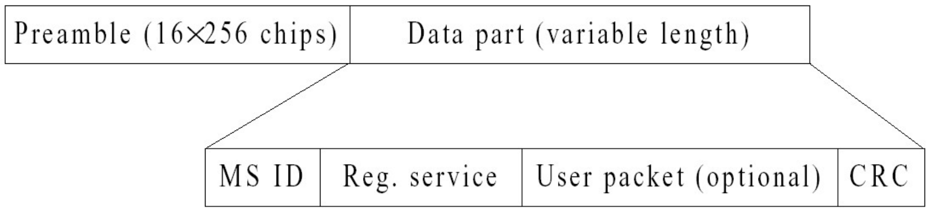
In addition, there are two dedicated channels:

- Dedicated control channel (DCCH) covers the two dedicated control channels: standalone dedicated channel (SDCCH) and associated control channel (ACCH)
- Dedicated traffic channel (DTCH) for point-to-point data transmission in uplink and downlink

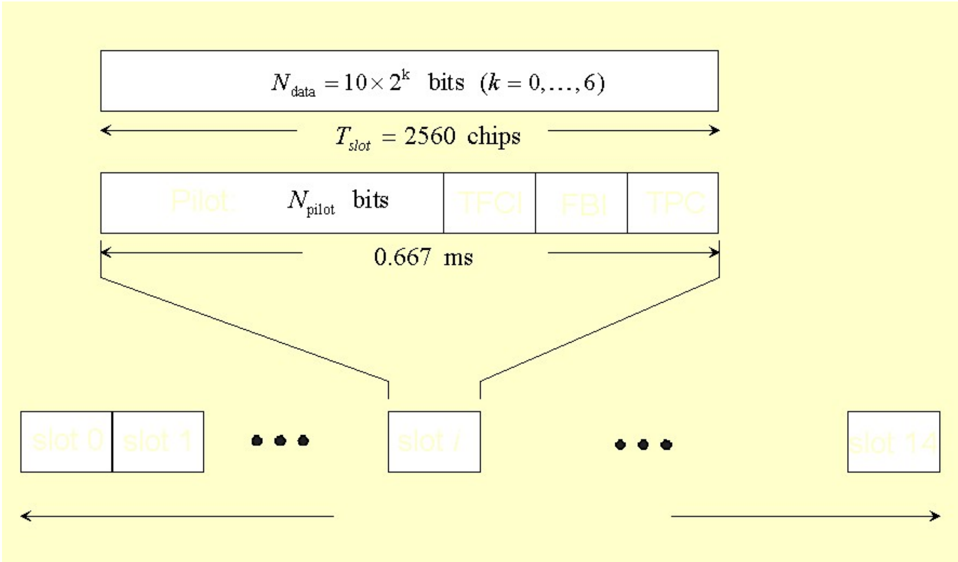
Uplink physical channel

There are two dedicated channels and one common channel on uplink.

- User data is transmitted on the dedicated physical data channel (DPDCH
- Control information is transmitted on the dedicated physical control channel (DPCCH)
- The random access channel is a common access channel (slotted ALOHA)



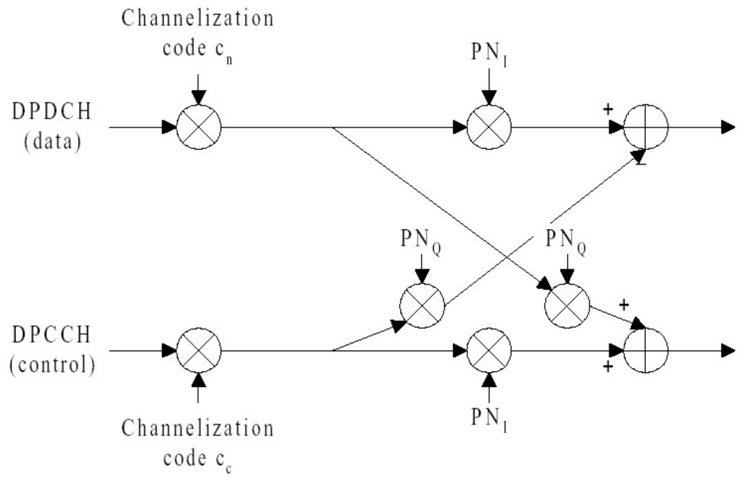
WCDMA Uplink Frame Structure



Each DPDCH frame on a single spreading code carries 10×2^k bits, where $k = 0$, 1, ..., 6, corresponding to a spreading factor of $256/2^k$

Multiple parallel variable rate services (i.e. dedicated logical traffic and control channels) can be time multiplexed within each DPDCH frame.

The overall DPDCH bit rate is variable on a frame-by-frame basis.



The uplink DPDCH and DPCCH are mapped onto the I and Q branches, respectively

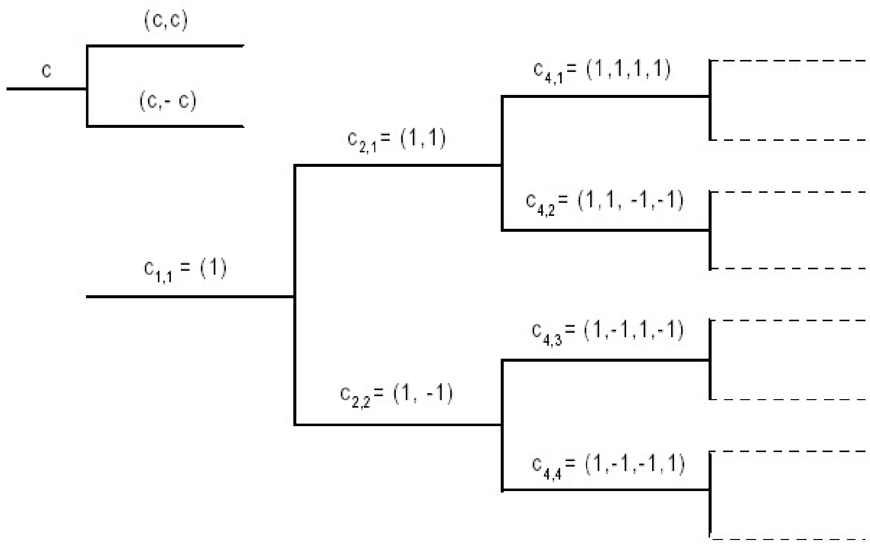
The I and Q branches are then spread to the chip rate with two different channelization codes c_n and c_c

Scrambled by a MS specific complex scrambling code $PN_I + jPN_Q$.

Rate Examples

- SF $= 256$; Chan. bit rate $= 15$ kbps (7.5 kbps)
- SF $= 128$; Chan. bit rate $= 30$ kbps (15 kbps)
- ...
- SF $= 4$; Chan. bit rate $= 960$ kbps (480 kbps)
- SF $= 4$; Chan. bit rate $= 5740$ (6 codes)

Orthogonal Variable Spreading Factor Codes



- For fixed chip rate, desired information rate determines length of spreading sequence and therefore processing gain.

- When a specific code is used, no other code on the path from that code to the root and or on the subtree beneath that code may be used.

Orthogonal Variable Spreading Factor Codes

- All the codes at any depth into the tree are the set of Walsh Sequences.

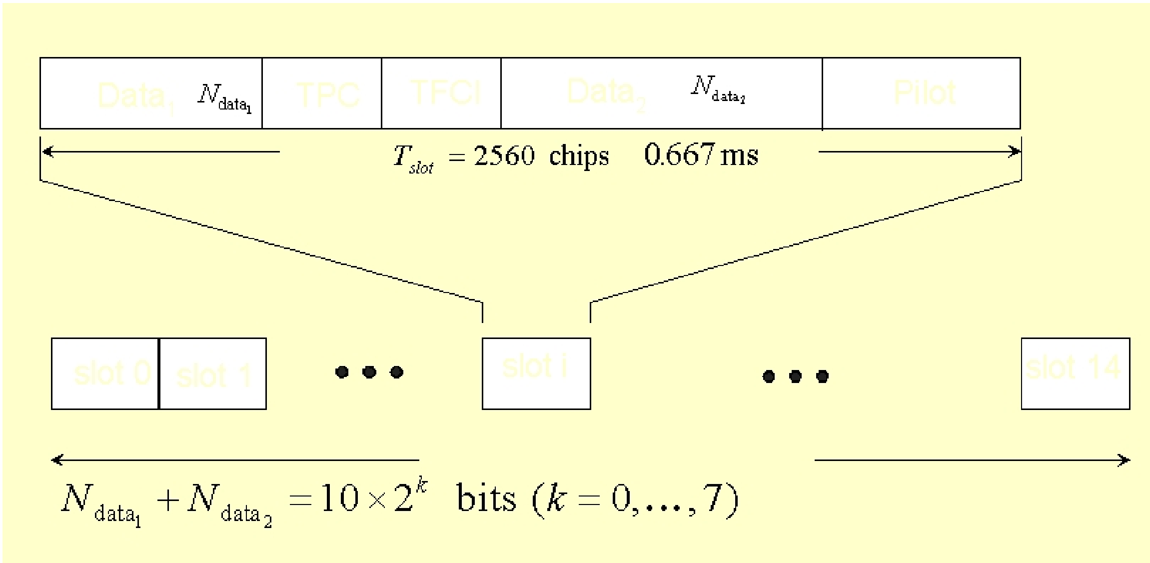
- Multicode used only for SF $= 4$

Downlink physical channels

There are three common physical channels

- Primary and secondary control physical channels (CCPCH): carry the downlink common control signals (BCCH, PCH and CCCH)
- SCH provides timing information and is used for handover measurements by the mobile stations
- Dedicated physical channels (DPDCH and DPCCH)- time multiplex

Downlink frame structure



- Max. raw data rate = 1280/0.67 ms = 1920 kbps, all slots
- Up to 3 parallel slots = 2.3 Mbps (rate 1/2 coding)
- Dedicated SF (does not change frame to frame)
- Use discontinuous transmission to accommodate rates

WCDMA Forward Error Control

Convolutional Coding

- rate 1/2 and rate 1/3
- 256 state
- puncture to higher rates
- interleave over 10, 20, 40 or 80 ms

Turbo Coding

- parallel coding rate 1/3
- 8 state codes
- block lengths 320 to 5114 bits

Packet Data

- Short data packets
  - Can be appended directly to a random access burst (common channel packet transmission)
  - used for short infrequent packets
  - the delay associated with a transfer to a dedicated channel is avoided
  - for common channel packet transmission only open loop power control is in operation
  - Common channel packet transmission should therefore be limited to short packets that only use a limited capacity.
- Larger or more frequent packets
  - transmitted on a dedicated channel
  - A large single packet is transmitted using a single-packet scheme where the dedicated channel is released immediately after the packet has been transmitted.
  - In a multipacket scheme the dedicated channel is maintained by transmitting power control and synchronization information between subsequent packets.

Handover

- Base stations in WCDMA need not be synchronized
- Asynchronous base station operation makes WCDMA soft handovers more complicated than soft handovers in synchronous CDMA networks
- MS measures observed timing differences of the downlink SCHs from the two BSs
- MS reports the timing differences back to the serving BS
- The timing of a new downlink soft handover connection is then adjusted with a resolution of one symbol (i.e., the dedicated downlink signals from the two base stations are synchronized with an accuracy of one symbol)
- That enables the mobile Rake receiver to collect the macro diversity energy from the two base stations

DECT (Digital Enhanced Cordless Telecommunication)

- standardized by ETSI (ETS 300.175-x) for cordless telephones
- standard describes air interface between base-station and mobile phone
- Characteristics
  - frequency: 1880-1990 MHz
  - channels: 120 full duplex
  - duplex mechanism: TDD (Time Division Duplex) with 10 ms frame length
  - multplexing scheme: FDMA with 10 carrier frequencies, TDMA with 2x 12 slots
  - modulation: digital, Gauian Minimum Shift Key (GMSK)
  - power: 10 mW average (max. 250 mW)
  - range: approx. 50 m in buildings, 300 m open space

Beyond 3G

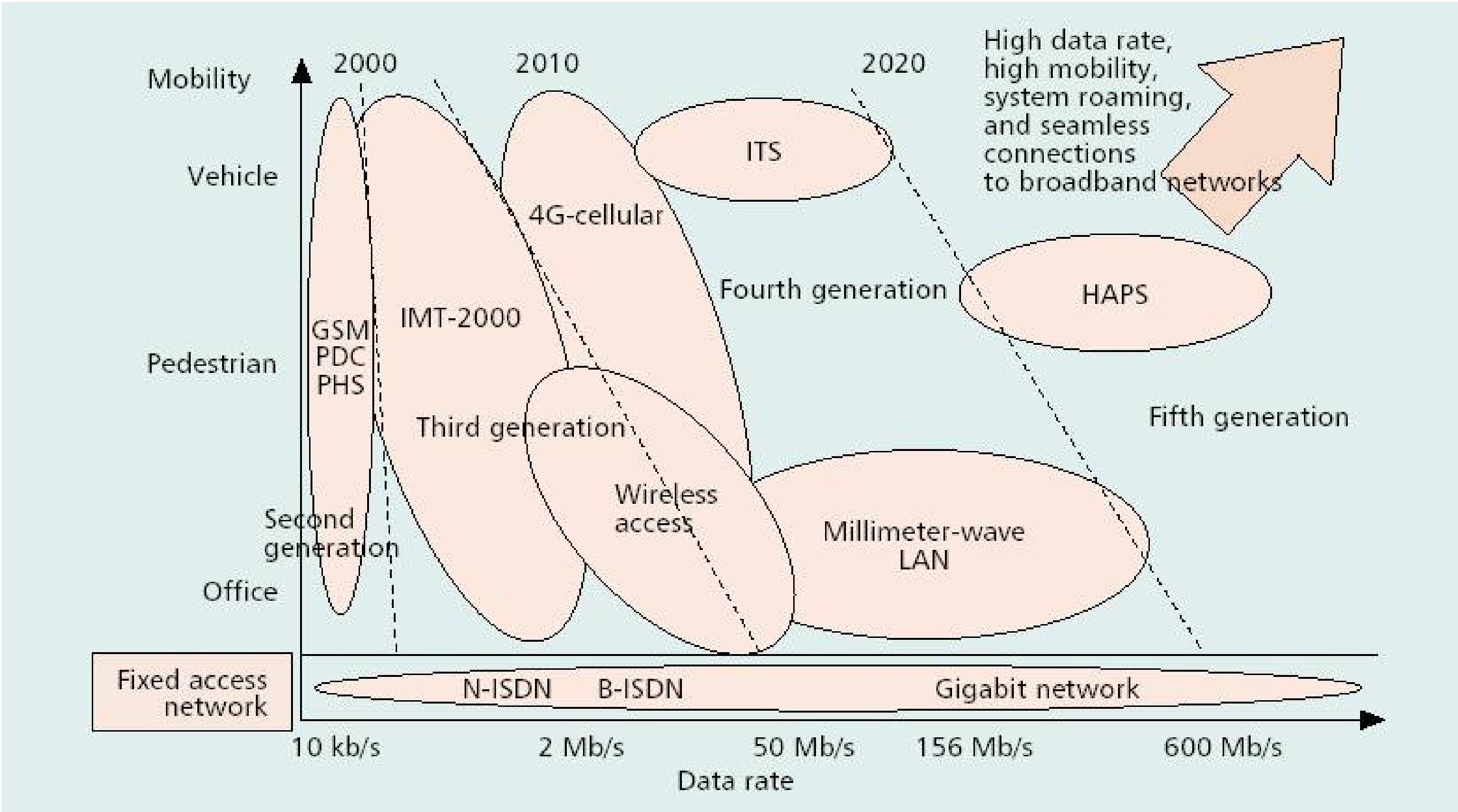


Figure 1:

Mobile broadband wireless access

Mission: The mission of IEEE 802.20 is to develop the specification for an efficient packet based air interface that is optimized for the transport of IP based services. The goal is to enable worldwide deployment of affordable, ubiquitous, always-on and interoperable multi-vendor mobile broadband wireless access networks that meet the needs of business and residential end user markets.

MBWA Scope: Specification of physical and medium access control layers of an air interface for interoperable mobile broadband wireless access systems, operating in licensed bands below 3.5 GHz, optimized for IP-data transport, with peak data rates per user in excess of 1 Mbps. It supports various vehicular mobility classes up to 250 Km/h in a MAN environment and targets spectral efficiencies, sustained user data rates and numbers of active users that are all significantly higher than achieved by existing mobile systems.

Digital Audio broadcasting

Digital audio broadcasting is an advance in radio broadcasting technology since the introduction of frequency modulated (FM) stereo radio

It provides high quality interference free audio reception to the user

The following principle features have been specified for DAB:

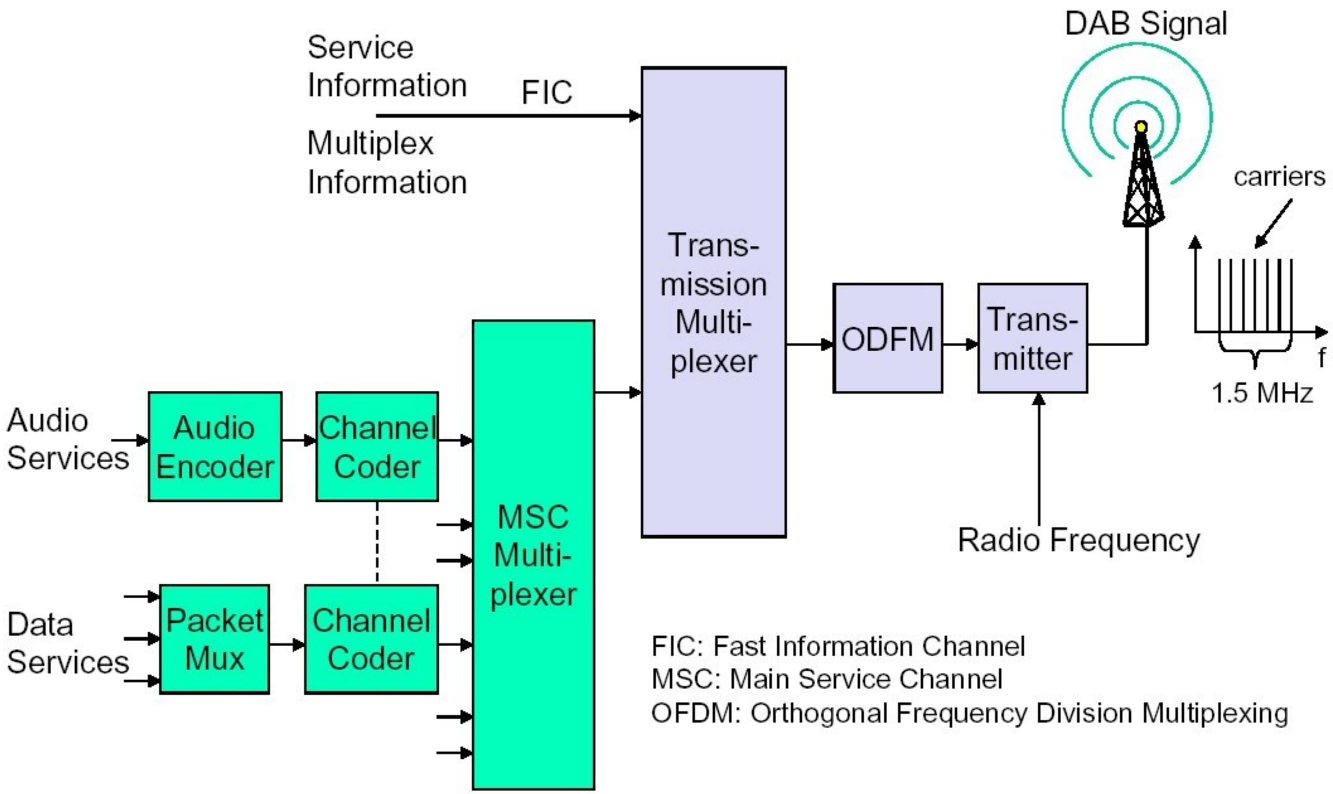
- Flexible audio coding schemes, from 8 kbps to 384 kbps, which allows the multiplexer to be configured in such a way that it provides typically 5 to 6 quality stereo audio programs or up to 20 restricted quality mono programmes.
- Data services, each service can be separately defined or further divided by means of a packet structure.
- Program associated data (PAD), embedded in the audio bit-stream for data transmitted with the audio programme.
- Conditional access (CA), applicable to each individual service or packet in the case of packet-mode data. This provides a scrambling mechanism.

- Service information (SI), used for operation and control of the receivers and provides information for programme selection to the user.

Table 1: DAB system parameters.

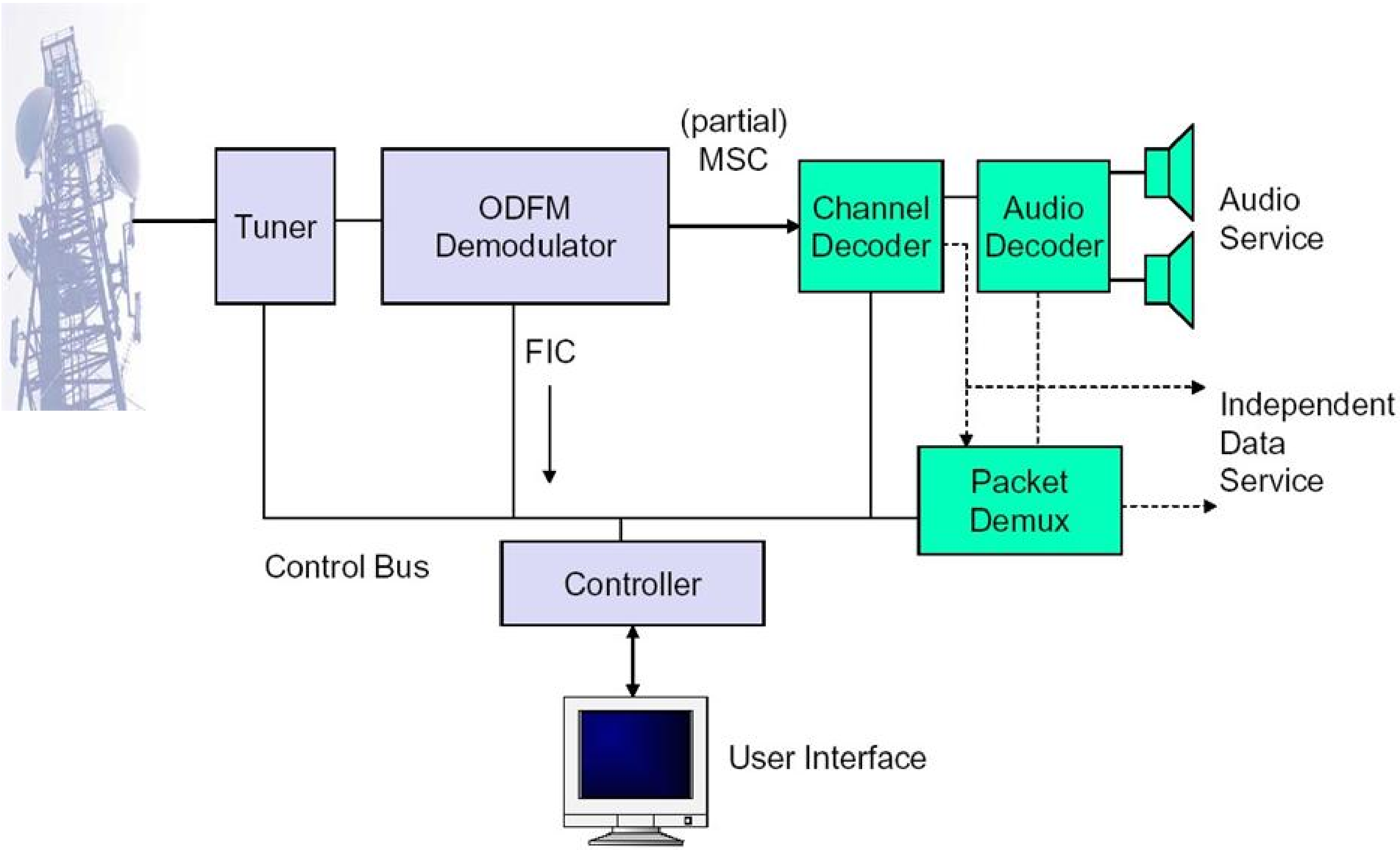
Parameters	Mode		
	I	II	III
Application	SFN	Terrestrial	Satellite
Modulation		Differential QPSK	
Total number of subcarriers	1536	384	192
OFDM symbol duration	1246 μs	312 μs	156 μs
Guard interval	246μ s	62μ s	31μ s
Frequency range	≤ 375 MHz	≤ 1.5 GHz	≤ 3 GHz

DAB Transmitter



(http://www.jochenschiller.de)

DAB receiver



(http://www.jochenschiller.de)

Digital video broadcasting

Two modes of operation are defined: “2K mode” and “8K mode”.

The “2K mode” is suitable for single transmitter operation and for small (single frequency networks) SFN networks with limited transmitter distances.

The “8K mode” can be used for single transmitter operation as well as for small and large SFN networks.

The system allows different levels of QAM modulation and different inner code rates to be used to trade bit rate versus ruggedness.

Table 2: DVB system parameters for "2K" mode.

Parameters	Value
Information data rate	5-30 Mbps
Modulation	QPSK,16QAM,64QAM
FEC code	Reed Solomon outer code
	Convolutional inner code
Code rates	1/2, 2/3,3/4
Total number of subcarriers	1705 (2K mode)
OFDM symbol duration	303 μs
Guard interval	75.9 μs
Signal bandwidth	5.62 MHz

UMTS/DVB/DAB

	UMTS	DAB	DVB
Spectrum bands (depends on national regulations) [MHz]	2000 (terrestrial), 2500 (satellite)	1140-1504, 220-228 (UK)	130-260, 430-862 (UK)
Regulation	Telecom, licensed	Broadcast, licensed	Broadcast, licensed
Bandwidth	5 MHz	1.5 MHz	8 MHz
Effective throughput	30-300 kbit/s (per user)	1.5 Mbit/s (shared)	5-30 Mbit/s (shared)
Mobility support	Low to high	Very high	Low to high
Application	Voice, data	Audio, push Internet, images, low res. video	High res. video, audio, push Internet
Coverage	Local to wide	Wide	Wide
Deployment cost for wide coverage	Very high	Low	Low

Broadband wireless access

- Allows simultaneous wireless delivery of voice, data, and video
- BW is considered a competing technology with Digital Subscriber Line (DSL)
- Requires clear line of sight between the transmitter and the receiving end
- Two categories: Local multi-point distribution service (LMDS) and Multi-channel multi-point distribution service (MMDS). Both operate in FCC-licensed frequency bands.
  - LMDS is a high bandwidth wireless networking service in the 28-31 GHz range of the frequency spectrum and has sufficient bandwidth to broadcast all the channels of direct broadcast satellite TV, all of the local over-the-air channels, and high speed full duplex data service. Average distance between LMDS transmitters is approximately 2 Km apart.
  - MMDS operates at lower frequencies, in the 2 GHz licensed frequency bands. MMDS has wider coverage than LMDS, up to 35 miles, but has lower throughput rates

IEEE 802.16 Wireless MAN

- A major new tool in the effort to link homes and businesses to core telecommunications networks worldwide.
- WirelessMAN technology bringing the network to a building, users inside the building will connect to it with conventional in-building networks such as, for data, Ethernet (IEEE Standard 802.3) or wireless LANs (IEEE Standard 802.11) and direct connection as well
- IEEE 802.16(2001) - 10-60GHz band:
- 802.16a(2003) 2-11GHz band

STC coded OFDM in IEEE 802.16a

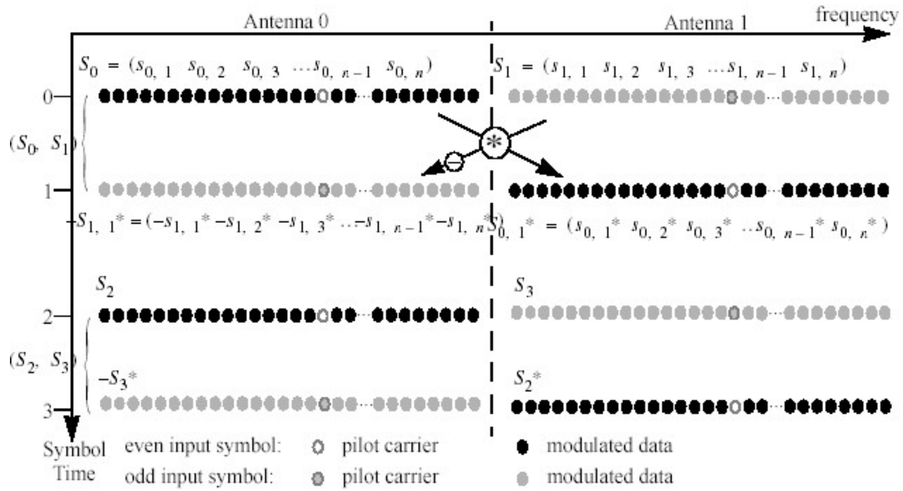
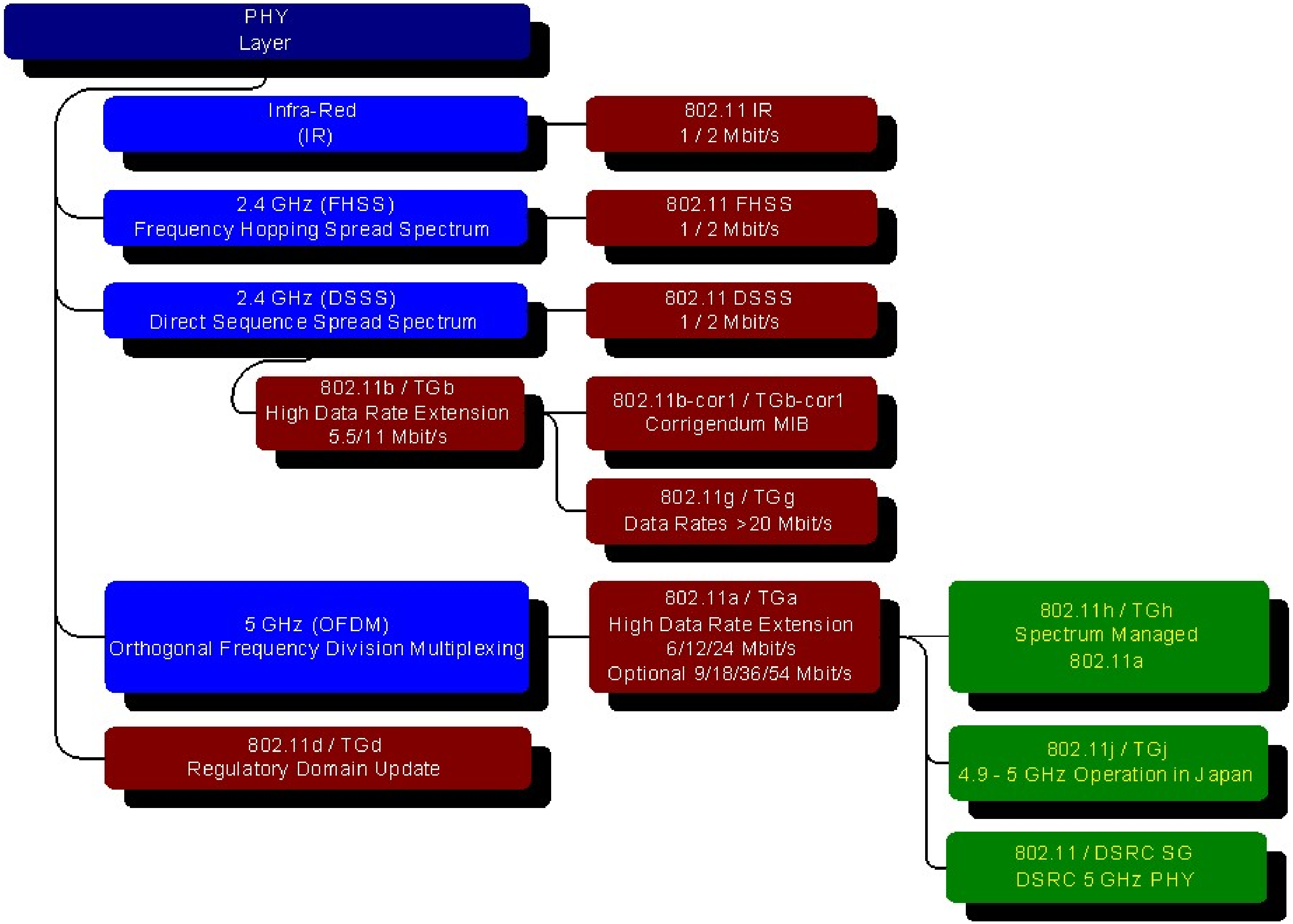


Figure 128aq—STC usage with OFDM

802.11 Activities

Task Group (TG)	Responsibilities
MAC	WLAN MAC in conjunction with PHY, CSMA/CA
PHY	WLAN PHY: IR, 2.4 GHz FHSS and DSSS, data rates of 1 & 2 Mbps
TGa	PHY for UNII (US 5 GHz), uses OFDM to obtain data rates from 6-54 Mbps. Almost the same as HiperLan2
TGb	Higher rate PHY for 2.4 GHz ISM-band, up to 11 Mbps. Uses CCK.
TGb_corl	Corrections to 802.11b
TGc	802.11 bridging
TGd	Operation in new regulatory domains, roaming.
TGe	QoS (previously also security, authentication). Improving capabilities and efficiencies to better service applications such as voice and video on wireless networks.
TGf	Inter-Access Point Protocol (IAPP) a higher layer protocol to allow roaming between multi-vendor access points
TGg	Higher data rates for the 2.4 GHz band, up to 54 Mb/s. Backwards compatible with 802.11b . Supports CCK and OFDM.
TGh	Use of 802.11a in 5 GHz band in Europe, includes dynamic frequency selection (DFS) and transmit power control (TPC)
TGi	Enhance MAC for security and authentication
TGj	802.11 and 802.11a PHY 5 GHz operation in Japan
TGk	Radio resource measurements (for higher layers)
TGl	-
TGm	802.11 standard corrections maintenance
TGn	High throughput PHY and MAC, by reduced overheads and higher data rates in the order of 108-320 Mbps. Scheduled for the year 2005-2006.

802.11 Activities - PHY:



Summary of the 802.11 PHY

Parameter	802.11 (FHSS)	802.11 (DSSS)	802.11b	802.11a	802.11g
Max MAC frame	4,095 bytes	4-8,191 bytes	4,095 bytes	4,095 bytes	4,095 bytes
Slot time	50 µs	20 µs	20 µs	9 µs	-
SIFS time[1]	28 µs	10 µs	10 µs	16 µs	-
Contention Window Size	15-1,023 slots	31-1,023 slots	31-1,023 slots	15-1,023 slots	-
Preamble Duration	96 µs	144 µs	144 µs	20 µs	-
PLCP Header duration	32 µs	48 µs	24 or 48 µs[2]	4 µs	-
Operation Band	2.4 GHz	2.4 GHz	2.4 GHz	5 GHz	2.4 GHz
Data Rates Supported	1 & 2 Mbps	1 & 2 Mbps,	1, 2, 5.5 & 11 Mbps	6*, 9, 12*, 18, 24*, 36 & 54 Mbps,	1*, 2*, 5.5*, 6, 9, 11*, 12*, 18, 24*, 36 & 54

802.11b Wi-Fi (wireless fidelity) (IEEE 802.11b)

- Operates in 2.4GHz-2.4835GHz frequency band
- Direct Sequence Spread Spectrum (DSSS)
- Three non-overlapping 22MHz channels
- Data Rates and Modulation/Coding schemes
  - 1Mbps – DBPSK (Differential Binary Phase Shift Keying)
  - 2Mbps – DQPSK (Differential Quadrature Phase Shift Keying)
  - 5.5Mbps – CCK (Complementary Code Keying)
  - 11Mbps – CCK (Complementary Code Keying)
- Nominal signal strength is 100mW (20dBm)
- Preamble and header always transmitted at 1Mbps

- The 802.11g standard runs on the same frequency as Wi-Fi but promises speeds up to 54 Mbps;

for CCK - "http://www.csee.wvu.edu/ jian/article/cck.htm"

IEEE 802.11a)

Table 3: IEEE 802.11a PHY parameters.

Parameters	Value
Information data rate	6,9,12,24,36 48 and 54 Mbps
Modulation	BPSK-OFDM QPSK-OFDM 16QAM-OFDM 64QAM-OFDM
FEC code	Convolutional rate 1/2, (K=7)
Code rates	1/2, 2/3,3/4
Total number of subcarriers	52
Number of pilot subcarriers	4
OFDM symbol duration	4 μs
Guard interval	0.8 μs
Signal bandwidth	16.6 MHz

HIPERLAN/II

IEEE 802.15 - WPAN TG1

- The IEEE Project 802.15.1 has derived a Wireless Personal Area Network standard based on the Bluetooth v1.1 Foundation Specifications
- The IEEE Std 802.15.1-2002 was published 14Jun02.

Usage model of Bluetooth

- Voice/data access point
- Peripheral Interconnects
- Personal Area Networking

Bluetooth Specifications, Bluetooth SIG at http://www.bluetooth.com/.

Bluetooth

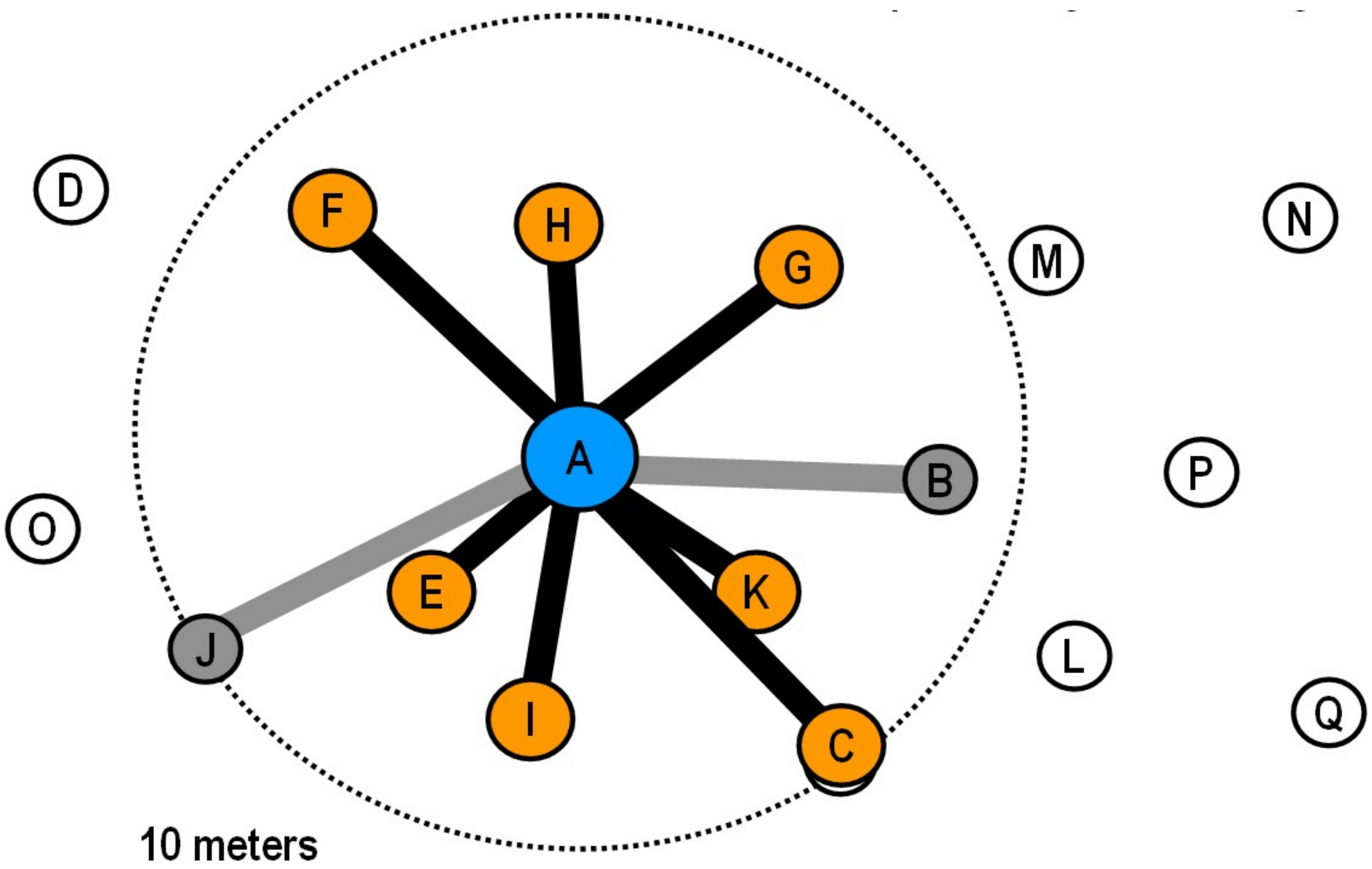
- This allows devices to communicate at speeds up to 1 Mbps within 30 feet.
- Operates in the 2.4 GHz ISM band (2400-2483.5 MHz) and uses FH-TDD
- FH Parameters
  - 79 carriers spaced at 1 MHz

$$f = 2402 + kMHz \;\; k = 0, \ldots, 78$$

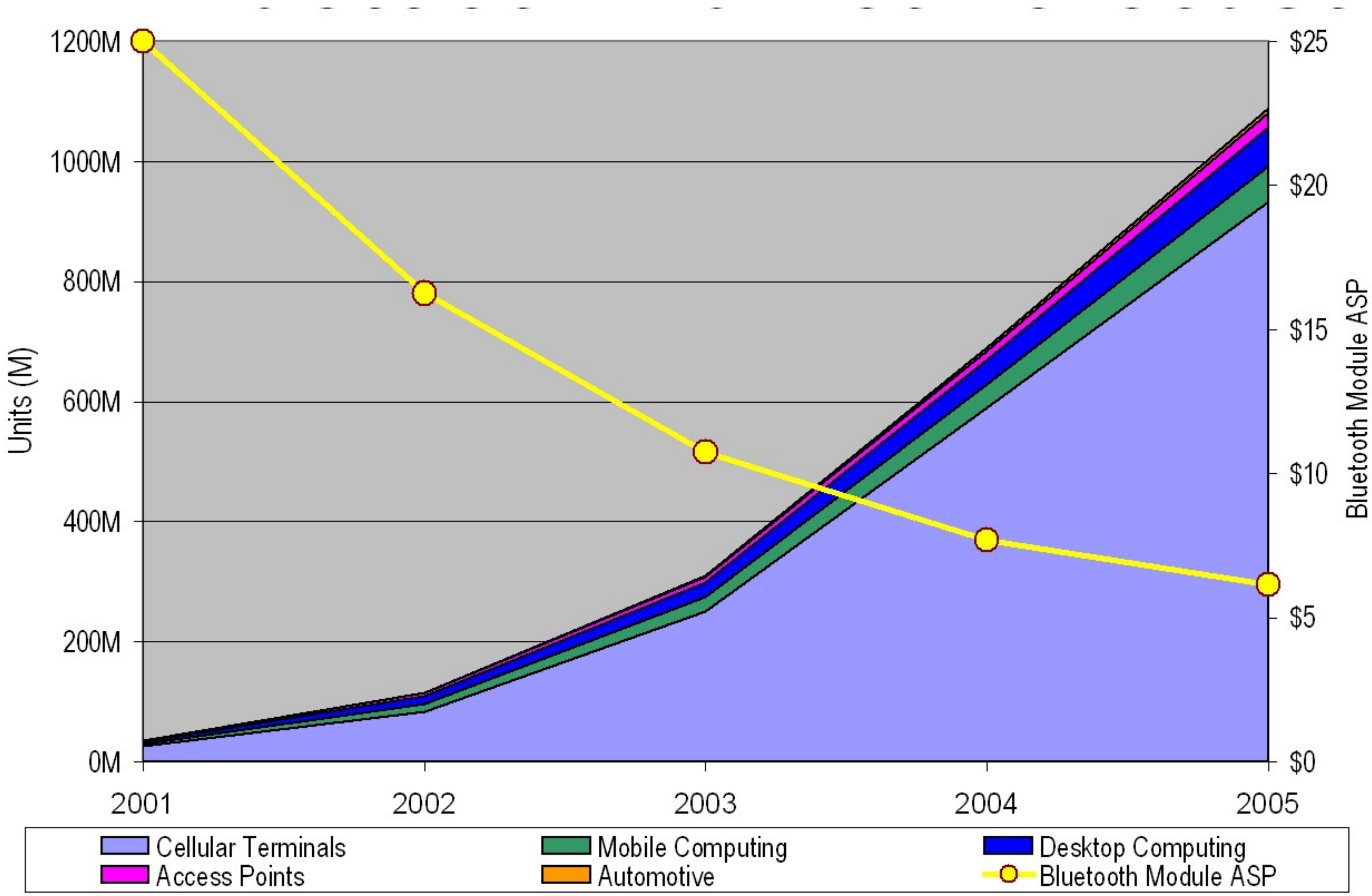
  - Nominal hopping rate 1600 hops per second
- Modulation parameters
  - FSK modulation, Gaussian shaping BT=0.5

Creating a piconet

- – Inquiry
  – Paging (7 active slaves)
  – Parking (Upto 256 slaves)



Bluetooth Market Forecast



IEEE 802.15 - WPAN TG3

The IEEE 802.15 High Rate Alternative PHY Task Group (TG3a) for Wireless Personal Area Networks (WPANs) is working to define a project to provide a higher speed PHY enhancement amendment to 802.15.3 for applications which involve imaging and multimedia.

Info. Data Rate	55 Mbps*	80 Mbps**	110 Mbps*	160 Mbps**	200 Mbps*	320 Mbps**	480 Mbps**
Modulation/Constellation	OFDM/QPSK	OFDM/QPSK	OFDM/QPSK	OFDM/QPSK	OFDM/QPSK	OFDM/QPSK	OFDM/QPSK
FFT Size	128	128	128	128	128	128	128
Coding Rate (K=7)	R = 11/32	R = 1/2	R = 11/32	R = 1/2	R = 5/8	R = 1/2	R = 3/4
Spreading Rate	4	4	2	2	2	1	1
Data Tones	100	100	100	100	100	100	100
Info. Length	242.4 ns	242.4 ns	242.4 ns	242.4 ns	242.4 ns	242.4 ns	242.4 ns
Cyclic Prefix	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns
Guard Interval	9.5 ns	9.5 ns	9.5 ns	9.5 ns	9.5 ns	9.5 ns	9.5 ns
Symbol Length	312.5 ns	312.5 ns	312.5 ns	312.5 ns	312.5 ns	312.5 ns	312.5 ns
Channel Bit Rate	640 Mbps	640 Mbps	640 Mbps	640 Mbps	640 Mbps	640 Mbps	640 Mbps
Multi-path Tolerance	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns	60.6 ns

http://www.multibandofdm.org/

Multiband OFDM

- Basic idea: divide spectrum into several 528 MHz bands.
- Information is transmitted using OFDM modulation on each band.
  - OFDM carriers are efficiently generated using an 128-point IFFT/FFT.
  - Internal precision is reduced by limiting the constellation size to QPSK.
- Information bits are interleaved across all bands to exploit frequency diversity and provide robustness against multi-path and interference.
- 60.6 ns cyclic prefix provides robustness against multi-path even in the worst channel environments.
- 9.5 ns guard interval provides sufficient time for switching between

Multiband OFDM Cntd..

- By using a contiguous set of orthogonal carriers, the transmit spectrum will always occupy a bandwidth greater than 500 MHz.
- Total of 128 tones:
  - 100 data tones used to transmit information (constellation: QPSK).
  - 12 pilot tones used for carrier and phase tracking.
  - 10 user-defined pilot tones.
  - Remaining 6 tones including DC are NULL tones.
- User-defined pilot tones:
  - Carry no useful information.
  - Energy is placed on these tones to ensure that the spectrum has a bandwidth greater than 500 MHz.
  - Can trade the amount of energy placed on tones for relaxing analog filtering specifications.
  - Ultimately, the amount of energy placed on these tones is left to the implementer. Provides a level of flexibility for the

IEEE 802.15 - WPAN TG4

- Investigate a low data rate solution with multi-month to multi-year battery life and very low complexity.

- It is intended to operate in an unlicensed, international frequency band.

- Potential applications are
  - sensors
  - interactive toys
  - smart badges
  - remote controls
  - home automation

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Comparing wireless technologies

Comparing Wireless Technologies (Roughly)

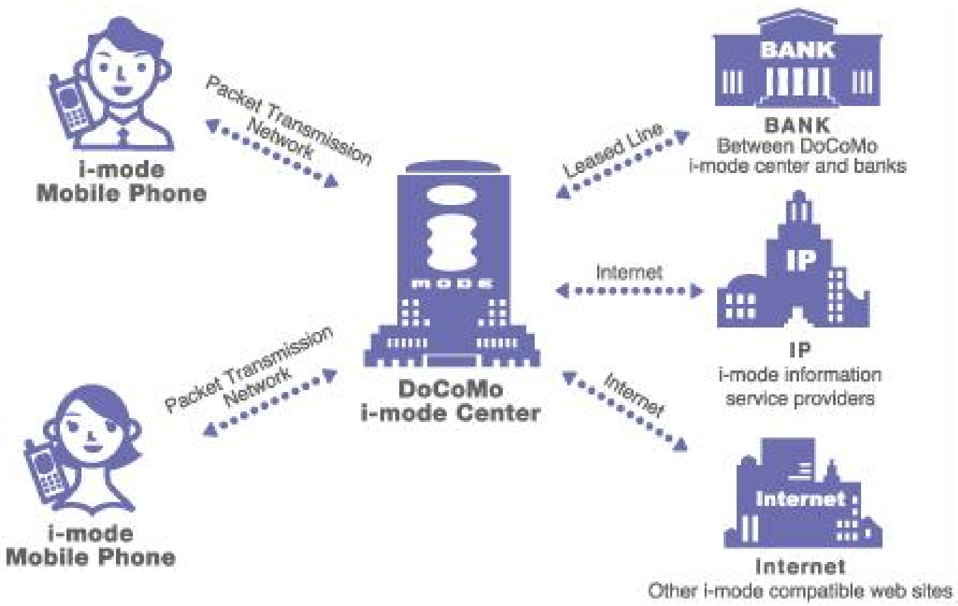
TECHNOLOGY	DATA RATE (Mb/s)	OUTPUT POWER (mW)	RANGE (meters)	FREQUENC BAND
Bluetooth	1-2	100	100	2.4 GHz
IrDA	4	100 mW/sr[1]	1-2	Infrared
Ultrawideband	100-500	1	10	3.1-10.6 GHz
IEEE 802.11a	54	40-800	20	5 GHz
IEEE 802.11b (Wi-Fi)	11	200	100	2.4 GHz
IEEE 802.11g	54	65	50	2.4 GHz

[1] Eye safety determines the infrared power density, which is measured in milliwatts per steradian

IEEE spectrum

i-Mode

- First introduced in Japan by NTT DoCoMO in 1999
- Offers wireless web browsing and e-mails from mobile phones
- Based on packet data transmission
- user is charged according to the volume of data transmitted



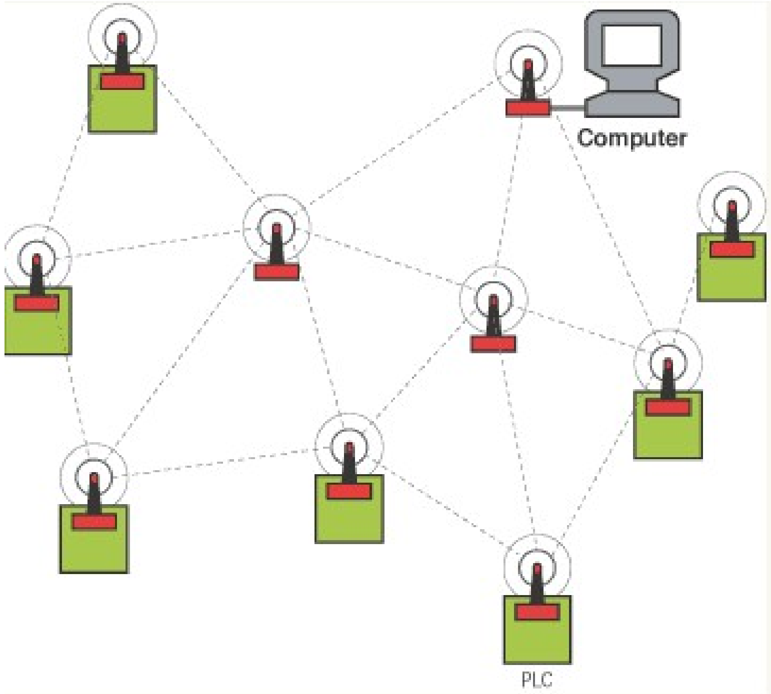
Free space optical

- High-speed Internet and intranet connections for the "last mile," without tearing up the streets to install cables.

- requires line-of-sight access from a transceiver, situated on the top of a building or inside near a window, to a central hub, which then attaches to a fiber-optic line.

- The laser beam is usually transmitted hundreds of feet in the air

Mesh networks



In a wireless mesh network, multiple nodes cooperate to relay a message to its destination.

Mobile adhoc networks

- MANET is an autonomous collection of mobile users that communicate over relatively bandwidth constrained wireless links
- Since the nodes are mobile, the network topology may change rapidly and unpredictably over time
- The network is decentralized, where all network activity including discovering the topology and delivering messages must be executed by the nodes themselves, i.e., routing functionality will be incorporated into mobile nodes.

Radio frequency identification (RF ID)

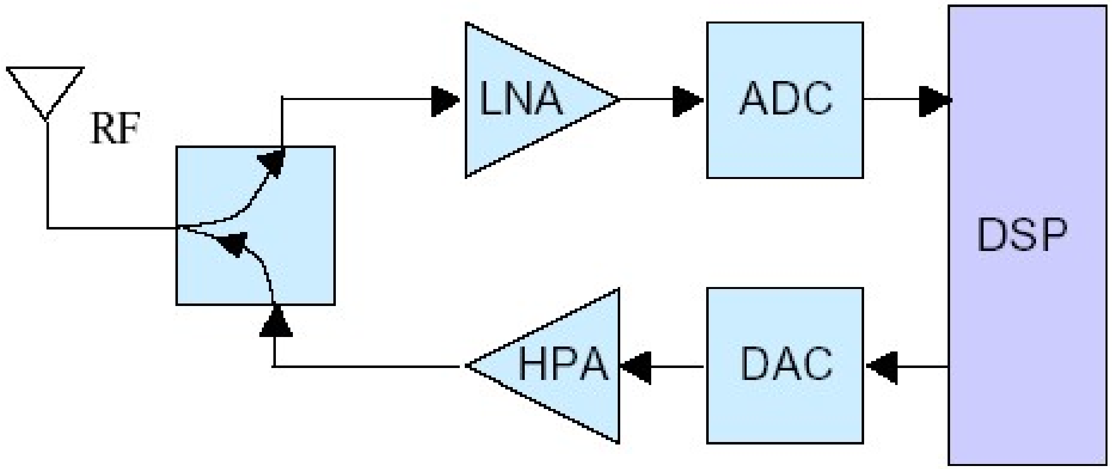
- Tiny tags containing a microprocessor, an antenna, and an identification code are embedded in store items
- Passive devices
- Special readers track the items for automated checkout and inventory keeping

Smart antennas

- Smart antennas combine multiple antenna elements (called an array) with a signal processing capability to transmit and receive in an adaptive manner, so that the system automatically changes the direction of the radiation/reception pattern in response to the position of the mobile user
- Smart Antenna systems can customize beam patterns for each mobile user by means of internal feedback control.
- Smart Antenna systems can be characterized under 2 major categories
  - Switched Beam: A finite number of fixed, predefined patterns or combining strategies
  - Adaptive Array: An infinite number of patterns (scenario based) that are adjusted in real time
- Smart antenna offers
  - Increased capacity, Extended range, Better link quality, longer battery life in the mobile unit, mobile user location estimation, communication security etc.

Multiple input Multiple output (MIMO) systems and Space time signal processing (space time coding)

Software defined radio



Software defined radio is a system which has an operating range, bandwidth, level of power, type of modulation, channel coding and so on, all of them configurable by software without changing the hardware