UNIT-II

Wireless Channel and Delay Spread:

Power Profile:-

The power profile of wireless communication system is

$$\phi(\tau) = |h(\tau)|^2$$

We Know that: $$h(\tau) = \sum_{i=0}^{L-1} a_i \delta(t - \tau_i)$$

$$\phi(\tau) *= \sum_{i=0}^{L-1} |a_i|^2 \delta(t - \tau_i)$$

Where $|a_i|^2$ = Arriving power

$$\phi(\tau) = \sum_{i=0}^{L-1} g_i \delta(t - \tau_i)$$

Where $g_i = |a_i|^2$

= Gain of the i^{th}path

Essentially, there is a series of signal copies that is arriving because of the scattering of the signal.

The first signal is essentially attenuated or amplified by 'a', which means it is transmitted power times magnitude of $a_0{}^2$, so the transmitted power is unity. So, the arriving signal having power magnitude $a_0{}^2$ which is g_0.

In the second path carrying a power $|a_1|^2 = g_1$ so on and so, forth until the $L - 1^{th}$ path carrying power $|a_{L-1}|^2 = g_{L-1}$

Consider an L=4 multipath channel

Gain	Delay
$|a_0|^2$	τ_0
$|a_1|^2$	τ_1
$|a_2|^2$	τ_2
$|a_3|^2$	τ_3

So, the arriving power profile in fading multipath channel is plotted as follows:

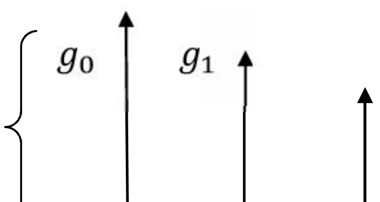
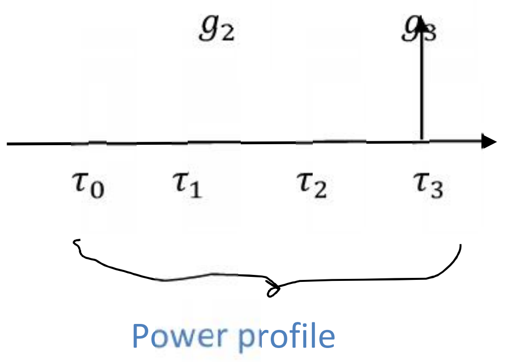


Fig: schematic of



an L=4 tap wireless

Channel profile

So, there are four paths. The O^{th} path is arriving at gain 0, after some delay another path arriving with some power gain1, after some delay (τ_2) at gain $g_2$and another path arriving at delay (τ_3) with power or gain at g_3. This is known as power profile. Because it represents the profile of the arriving signal copies.

This plot represents that power profile is a spread over which the signal copies are arriving with different delays and different powers.

Hence this is known as the power profile of the wireless communication system.

DELAY SPREAD:

Another sample power profile is shown below.

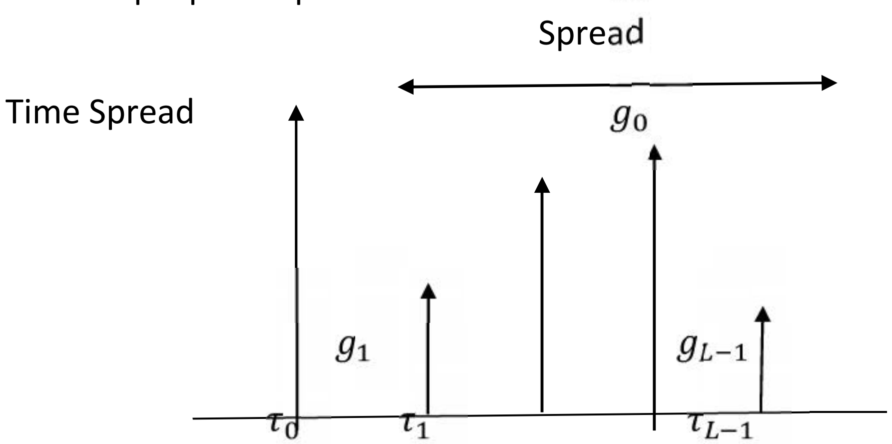


Fig: Schematic of a typical wireless channel power profile and delay spread

The profile consists of some path arriving with some power and so on. All these paths are corresponding to the same copies, which are arising due to the scatter components of the multipath propagation environment which means we get multiple signal copies except with delay because the signal has to propagate through different distances.

So, the first one is arriving at a delay τ_0, the second copy is arising at a delay τ_1 and so on and so forth until the final copy is arriving at some delay. There is also going to be an attenuation or amplification because of propagation losses as well as scattering losses. Multiple signal copies are arriving at not a single time, but over an interval of time. This time spread is termed as "**DELAY SPREAD**". This is denoted by σ_Z

- The total power received in a multipath wireless channel occurs over a spread of time referred to as the "**DELAY SPREAD**".

Maximum Delay Spread (σ_τ^{max}):-

A framework to quantify the delay spread of a wireless channel is through the maximum delay spread of the channel denoted by σ_T^{max}.

Consider a wireless channel with L multipath components, the first path i.e., first arriving component arrive at $\tau_0(delay)$.

The last arriving component arrives at delay τ_{L-1}due to scattering the delay spread is the delay between the first and the last arriving components.

Therefore the maximum delay spread is simply defined as

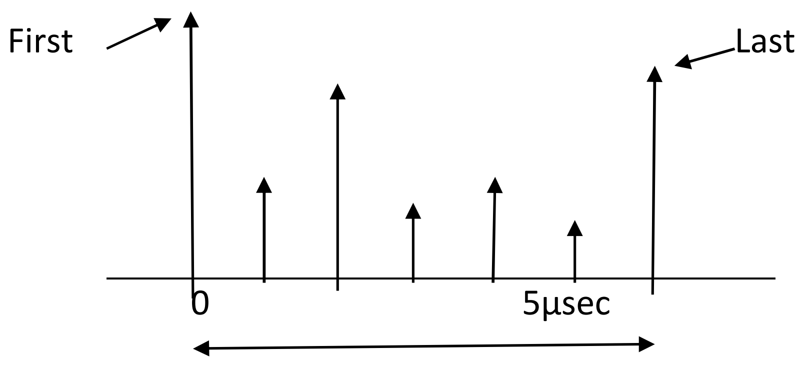
$$\sigma_\tau^{max} = \tau_{L-1} - \tau_0$$

- For instance, 4 path channel.
  Consider a system in which the first arriving signal is arriving at delay τ_0 =0µsec and the last arriving path which is the third path is arriving. So, the last path is arriving at a delay $\tau_{L-1} = \tau_{4-1}$=τ_3=5µsec.

- Then maximum delay spread

$$\begin{aligned} \sigma_\tau^{max} &= \tau_{L-1} - \tau_0 \\ &= \tau_3 - \tau_0 \\ &= 5\mu sec - 0\mu sec \\ &= 5\ \mu sec. \end{aligned}$$

A plot of this instance is shown below



Spread or maximum Spread

Fig: Power Profile

$$\begin{aligned} \sigma_\tau^{max} &= \tau_{L-1} - \tau_0 \\ &= 5-0 \\ &= 5\mu sec. \end{aligned}$$

Therefore Maximum spread, ${\sigma_\tau}^{max}$ =5µsec.

RMS Delay Spread:-

- In typical wireless channels, the paths which arrive later are significantly lower in power due to the larger propagation distances and weather reflections as shown in figure. This results in a large value of the maximum delay spread $\boldsymbol{{\sigma_\tau}^{max}}$ even though several of the later paths comprise weaker scatter components with negligible power.
- Thus, the maximum delay spread metric is not a reliable indicator of the true power spread of the arriving multipath signal components in such scenarios, since it does not weight the delays in proportion to the signal power in the multipath components.
- For this purpose, the RMS delay spread is a more realistic indicator of the spread of the signal power in the arriving components.

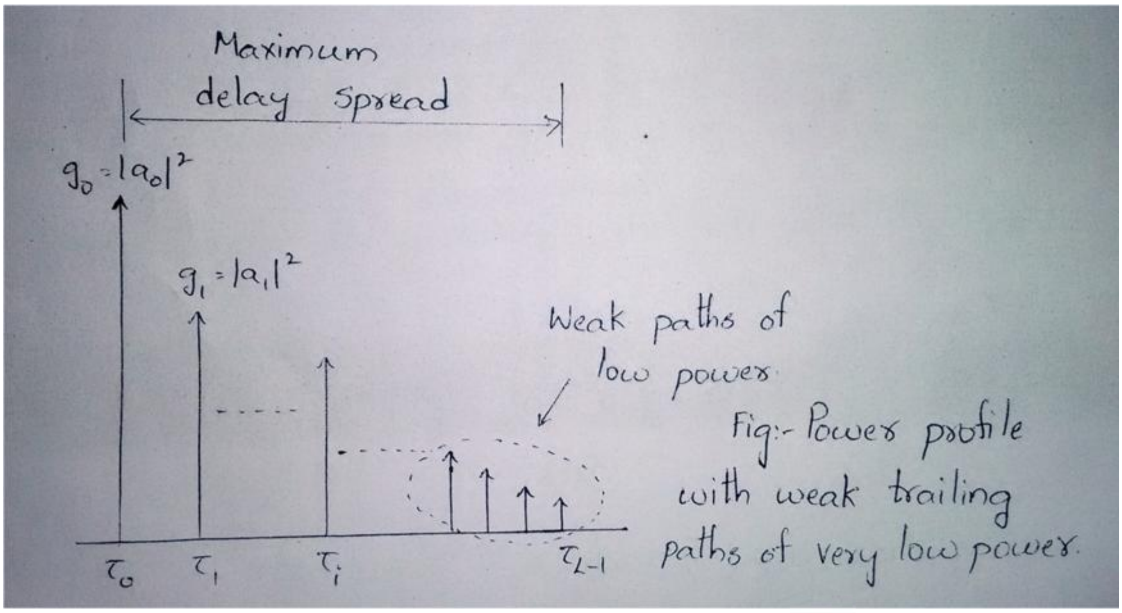


Fig:- Power profile with weak trailing paths of very low power.

- The acronym RMS stands for Root Mean Square delay spread. RMS is a term used to measure a sinusoidal signal.

  So, $\mathrm{g_i} = |\mathrm{a_i}|^2$ associated with delay τ_i
- If we have 'L' paths, then

  $\mathrm{g_0} = |\mathrm{a_0}|^2$ With τ_0

  $\mathrm{g_1} = |\mathrm{a_1}|^2$ With τ_1

  .

  .

  $\mathrm{g_{L-1}} = |\mathrm{a_{L-1}}|^2$ With τ_{L-1}
- The fraction of the power in the i^{th} path is given as

$$\mathrm{b_i} = \frac{\mathrm{g_i}}{\mathrm{g_0} + \mathrm{g_1} + - - - + \mathrm{g_{L-1}}}$$

g_i - i^{th} Power

$g_0 + g_1 + - - - + g_{L-1}$ – Total Power

$$= \frac{g_i}{\sum_{j=0}^{L-1} g_j}$$

- The average delay is given as

$\bar{\tau} = b_0\tau_0 + b_1\tau_1$ +---+$b_{L-1}\tau_{L-1}$

Fractional Power   Delay of Path

$$\bar{\tau} = \sum_{i=0}^{L-1} b_i\tau_i$$

$$= \sum_{i=0}^{L-1} \left(\frac{g_i}{\sum_{j=0}^{L-1} g_j}\right)\tau_i$$

$$= \sum_{i=0}^{L-1} \frac{g_i\tau_i}{\sum_{j=0}^{L-1} g_j}$$

Average Delay $$\bar{\boldsymbol{\tau}} = \frac{\sum_{i=0}^{L-1} g_i\tau_i}{\sum_{j=0}^{L-1} g_j}$$

- The delay spread is given by

$\sigma_\tau^{\ 2} = b_0(\tau_0 - \bar{\tau})^2 + b_1(\tau_1 - \bar{\tau})^2$ +---+$b_{L-1}(\tau_{L-1} - \bar{\tau})^2$

$(\sigma_\tau^{RMS})^2 = \sum_{i=0}^{L-1} b_i(\tau_i - \bar{\tau})^2$

$$= \sum_{i=0}^{L-1} \left(\frac{g_i}{\sum_{j=0}^{L-1} g_j}\right)(\tau_i - \bar{\tau})^2$$

$(\sigma_\tau^{RMS})^2 = \frac{\sum_{i=0}^{L-1} g_i(\tau_i - \bar{\tau})^2}{\sum_{j=0}^{L-1} g_j}$

Therefore the RMS delay spread is given by

$$\sigma_\tau^{max} = \sqrt{\frac{\sum_{i=0}^{L-1} g_i(\tau_i - \bar{\tau})^2}{\sum_{j=0}^{L-1} g_j}}$$

$\sum_{i=0}^{L-1} g_i(\tau_i - \bar{\tau})^2 \longrightarrow$ Average Square Deviation

$\sum_{j=0}^{L-1} g_j \longrightarrow$ Total Power

We know that

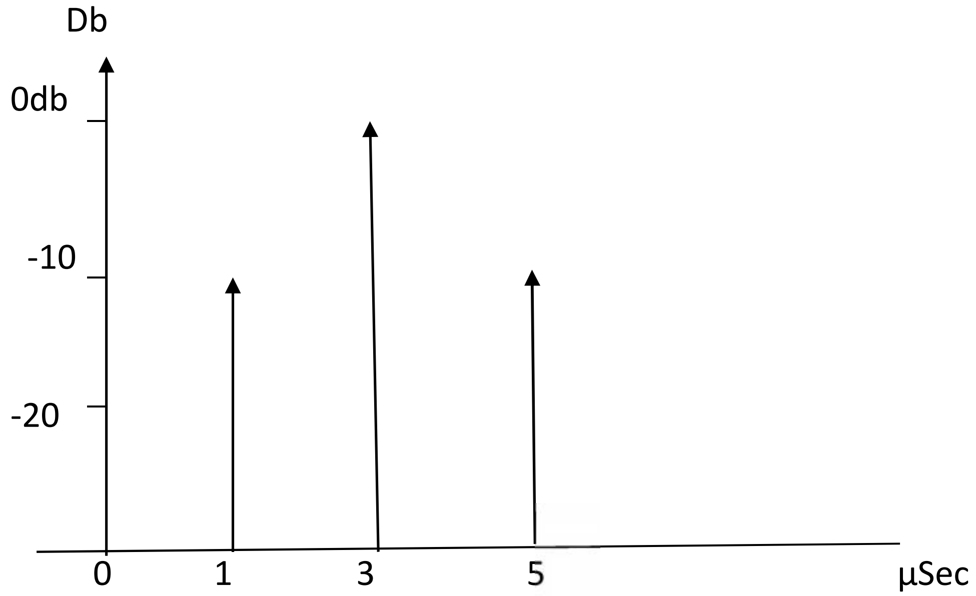
$$g_i = |a_i|^2$$

$$(\sigma_\tau^{RMS})^2 = \sqrt{\frac{\sum_{i=0}^{L-1}|a_i|^2\,(\tau_i-\bar{\tau})^2}{\sum_{j=0}^{L-1}|a_j|^2}}$$

RMS delay spread of the Wireless channel

Example:-

- Consider a channel in which there are four multipath, there are components i.e., L=4 Components.



- Consider the first path corresponding to τ_0 = 0µsec. The power associated with this path is g(db)= 20db.
- Hence, the linear power can be obtained as

$10\log_{10} g_0$ = -20db

$\log_{10} g_0$ = 2

$g_0 = 10^{-2}$

g_0 = 0.01

- Also, the amplitude a_0 associated with this path can be derived as

$a_0 = \sqrt{g_0} = \sqrt{0.01}$

$= \sqrt{10^2}$

$= \sqrt{\frac{1}{10^2}}$

$$= \frac{1}{10}$$

a_0 = 0.1

- Similarly, the gains of remaining paths are calculated and are listed in the table.

y, the g τ	Db(Gain)	**g**	a = listed in $\sqrt{g}$
0µsec	-20db	0.01	0.1
1 µsec	-10db	0.1	0.3162
3 µsec	0db	1	1
5 µsec	-10db	0.1	0.3162

- Now, compute the mean delay $\bar{\tau}$ for this channel as

$$\bar{\tau} = \frac{\sum_{i=0}^{4-1} g_i \tau_i}{\sum_{i=0}^{4-1} g_i} = \frac{\sum_{i=0}^{3} g_i \tau_i}{\sum_{i=0}^{3} g_i}$$

$$= \frac{g_0\tau_0 + g_1\tau_1 + g_2\tau_2 + g_3\tau_3}{g_0 + g_1 + g_2 + g_3}$$

$$= \frac{(0.01*0)+(0.1*1)+(1*3)+(0.1*5)}{0.01+0.1+1+0.1}$$

$$= \frac{0 + 0.1 + 3 + 0.5}{1.21}$$

$$= \frac{3.6}{1.21}$$

$\bar{\tau}$ = 2.9752μsec ⟶ Average weighted delay

- Delay Spread,

$$\sigma_\tau^2 = \frac{\sum_{i=0}^{L-1} g_i(\tau_i - \bar{\tau})^2}{\sum_{i=0}^{L-1} g_j}$$

- RMS delay spread,

$$\sigma_\tau = \sqrt{\frac{\sum_{i=0}^{4-1} g_i(\tau_i - \bar{\tau})^2}{\sum_{i=0}^{4-1} g_i}}$$

$$= \frac{(0.01*(0-2.9752)^2+0.1*(1-2.9752)^2+1*(3-2.9752)^2+0.1*(5-2.9752)^2}{0.01+01+1+0.1}$$

$$= (\frac{0.8893}{1.21})^2$$

Therefore RMS delay spread σ_τ = 0.8573µsec

Maximum delay spread $\sigma_\tau{}^{max} = \tau_{L-1} - \tau_0$

$$= \tau_{4-1} - \tau_0$$

$$= \tau_3 - \tau_0$$

=5µsec - 0 µsec

$$\sigma_\tau{}^{\max} = 5\mu\text{sec}$$

Therefore $\sigma_\tau{}^{RMS} < \sigma_\tau{}^{max}$

0.8573 < 5 µsec

- The RMS delay spread is much smaller than the maximum delay spread because the maximum delay spread id simply looking at the first and last components.
- However, many of these components carry an insignificant amount of power. So, the actual time interval over which most of the power is concentrated is much smaller than the time interval.
- So, this maximum delay spread is essentially a kind of a pessimistic, this gives equal weight to all the arriving multipath components.
- Here, RMS delay spread is a weighted combination of the delays of the different of delay spreads corresponding to different components. Hence, this is much smaller because the average duration over which the receiving power in the multipath wireless channel is smaller
- Therefore the RMS delay spread is a more appropriate number which weighs the different delays.

Average Power Profile:

- Consider the instantaneous power $|h(\tau)|^2$ corresponding to the delay τ. The average power associated with this delay can be defined as

$$\varphi(\tau) = |h(\tau)|^2$$

$$\bar{\varphi}(\tau) = \mathrm{E}|h(\tau)|^2$$

Average power profile

- The above quantity $\varphi(\tau)$ can be thought of as the average power associated with delay τ at various instants of time. It can also be thought of as the power at delay τ for wireless channels of different users in area.
- The former is averaging over time, while the latter represents an averaging over the ensemble of channels.
- Similar to the framework, we can define the fractional power associated with delay τ as

$$f(\tau) = \frac{\bar{\varphi}(\tau)}{\int_0^{\infty} \bar{\varphi}(\tau) d\tau}$$

$\bar{\varphi}(\tau)$ = Power at τ

F (τ) = Fractional power received at delay τ

$\int_0^{\infty} \bar{\varphi}(\tau) d\tau$ = Total power

- Where f (τ) denotes the power distribution density corresponding to the delay τ, i.e., f(τ) $\Delta\tau$ is the fraction of power in a delay interval of $\Delta\tau$ around τ.
- The average $\bar{\tau}$ can be defined as

$\bar{\tau} = \int_0^{\infty} \tau \, f(\tau) d\tau$

Average delay Delay Average fractional power

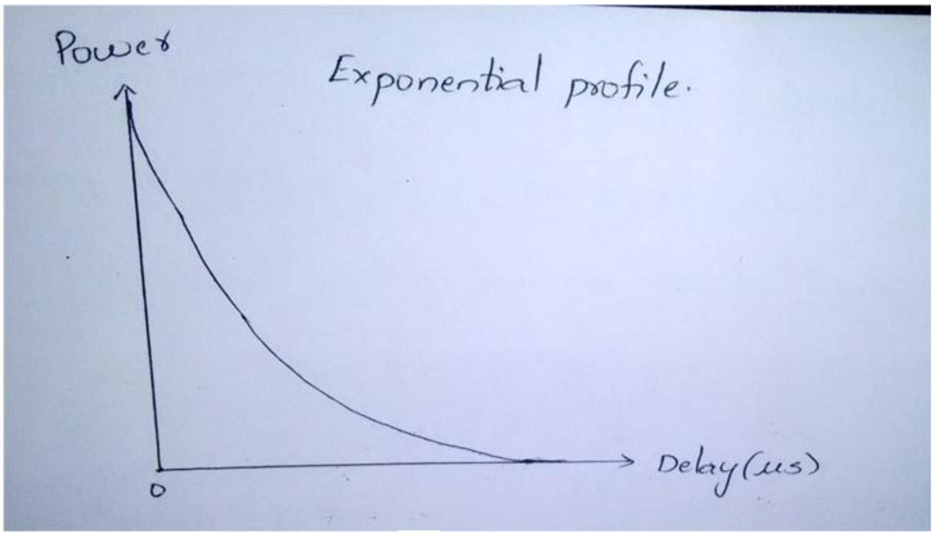
- Finally, the RMS delay spread for the above profile $\varphi(\tau)$ is defined as

$$\sigma_{\tau}^{max} = \sqrt{\int_0^{\infty} (\tau - \bar{\tau})^2 \, f(\tau) d\tau}$$

$$\sigma_{\tau}^{max} = \sqrt{\frac{\int_0^{\infty} (\tau - \bar{\tau})^2 \varphi(\tau) d\tau}{\int_0^{\infty} \varphi(\tau) d\tau}}$$

- Example:

Consider the average power profile $\varphi(\tau) = \alpha \, e^{\frac{-\tau}{\beta}}$, where α = 3db, β = 1μsec. compute the RMS delay spread σ_{τ}^{RMS} for this profile which is schematically shown in the below figure.



- Solution:
  α = 3db
  $$10 \log_{10}\alpha = 3$$
  $$\log_{10}\alpha = 0.3$$
  $\alpha = 10^{0.3}$
  $=1.99$
  $$\alpha = 2$$
   $\varphi(\tau) = \alpha\, e^{\frac{-\tau}{\beta}}$
  $= 2\, e^{\frac{-\tau}{\beta}}$
- The fractional power profile $f(\tau)$ can be obtained as
  $f(\tau) = \frac{\varphi(\tau)}{\int_0^{\infty} \varphi(\tau) d\tau}$
  $\int_0^{\infty} \varphi(\tau) = \int_0^{\infty} 2e^{\frac{-\tau}{\beta}} d\tau$
  $$= 2\beta e^{\frac{-\tau}{\beta}}$$
  $$= 2\beta$$
   $f(\tau) = \frac{2e^{\frac{-\tau}{\beta}}}{2\beta} = \frac{1}{\beta} e^{\frac{-\tau}{\beta}}$
- The average delay $\bar{\tau}$ is given as

$$\bar{\tau} = \int_0^\infty \tau f(\tau)\, d\tau$$

$$= \int_0^\infty \tau \left(\frac{1}{\beta} e^{-\tau/\beta} \right) d\tau$$

$$= \beta e^{-\tau/\beta} \Big|_0^\infty + \int_0^\infty e^{-\tau/\beta}\, d\tau$$

$$= \beta e^{-\tau/\beta} \Big|_0^\infty$$

$$= \beta$$

$$\bar{\tau} = 1 \mu s$$

- RMS delay spread:

  $$\sigma_\tau^{\,2} = \int_0^\infty (\tau - \bar{\tau})^2 f(\tau)\, d\tau$$

  $$= \int_0^\infty (\tau - \beta)^2 \frac{1}{\beta} e^{\frac{-\tau}{\beta}}\, d\tau$$

  $$= \beta^2$$

  $$\sigma_\tau = \sqrt{\beta^2} = \beta = 1\mu\text{sec}$$

  RMS delay spread, σ_τ = 1µsec

- The power profile looks like a decaying exponential as a function of the delay.

Average spread delay:

- Consider an outdoor wireless communication scenario. The cell radii of typical cells are in the range of 1-5km i.e., outdoor wireless signal. Propagation distances are of the order of few kms.

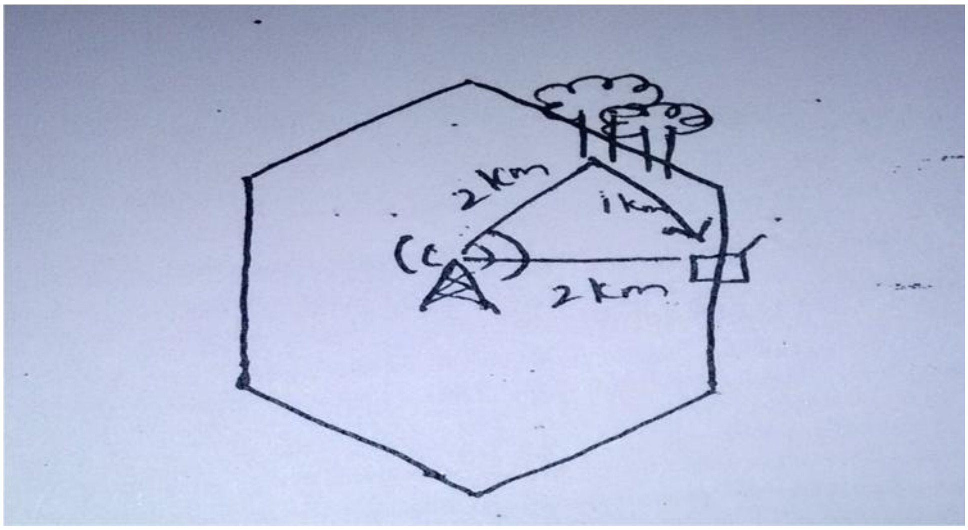


Fig: Typical delay spread in outdoor cellular channels.

- Consider two paths illustrated in figure, where the direct and scatter distances are given as d_0= 2km, d_1= 3km respectively. Hence the propagation delays τ_0, τ_1 are given as

$\tau_0 = \frac{2km}{c}$, $\tau_1 = \frac{3km}{c}$

Where C= $3X10^8$ m/s

- Delay spread, $\sigma_\tau^{max} = \Delta\tau$

$$= \tau_1 - \tau_0$$
$$= \frac{\Delta d}{c}$$
$$= \frac{3000m - 2000m}{3X10^8}$$
$$= \frac{1000}{3X10^8}$$
$$\sigma_\tau^{max} = 3.33\mu sec$$

 Average outdoor delay spreads are around 1-3µsec. Also, similarly corresponding to outdoor distances of around 10m, typical indoor delay spreads are of 10-50ns.

COHERENCE BANDWIDTH IN WIRELESS COMMUNICATION:

The important parameters of a wireless communication channel are the coherence bandwidth 'B_c'.

The channel delay profile is h (τ).

The frequency response H (f) of the wireless channel as

$$H(f) = \int_0^{-\infty} H(f)e^{-i2\pi ft}\, dt$$

Consider a simple case corresponding to sigma τ=0. In this scenario, since the delay spread is zero, the wireless channel comprises a single propagation path. Hence, the delay profile h(τ) is given as

$$H\ (\tau) = \delta\ (\tau)$$

The frequency response H(f) is given as

$$H(f) = \int_0^{\infty} a0\ \delta(\tau)e^{-i2\pi\tau}d\tau = 1$$

Thus the frequency response is the constant 1 and |H(f)|=1. This is basically a flat frequency response over the entire frequency band as shown in figure i.e., of infinite bandwidth .

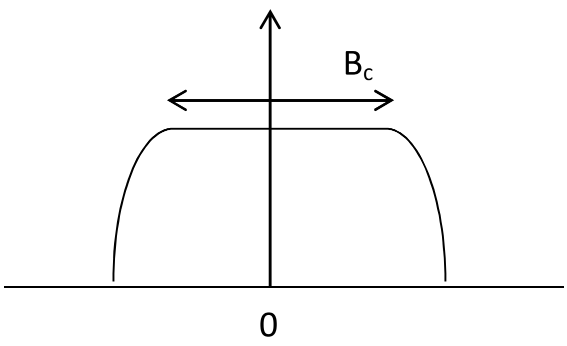
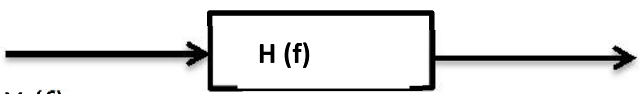


Fig: coherence bandwidth of wireless channel response

The response is approximately constant for bandwidth around 0 then it starts following already it has a some kind of a low pass characteristics. So, it is flat i.e., it starts falling alright and this portion of the spectrum over which the response is approximately constant is known as the coherence bandwidth

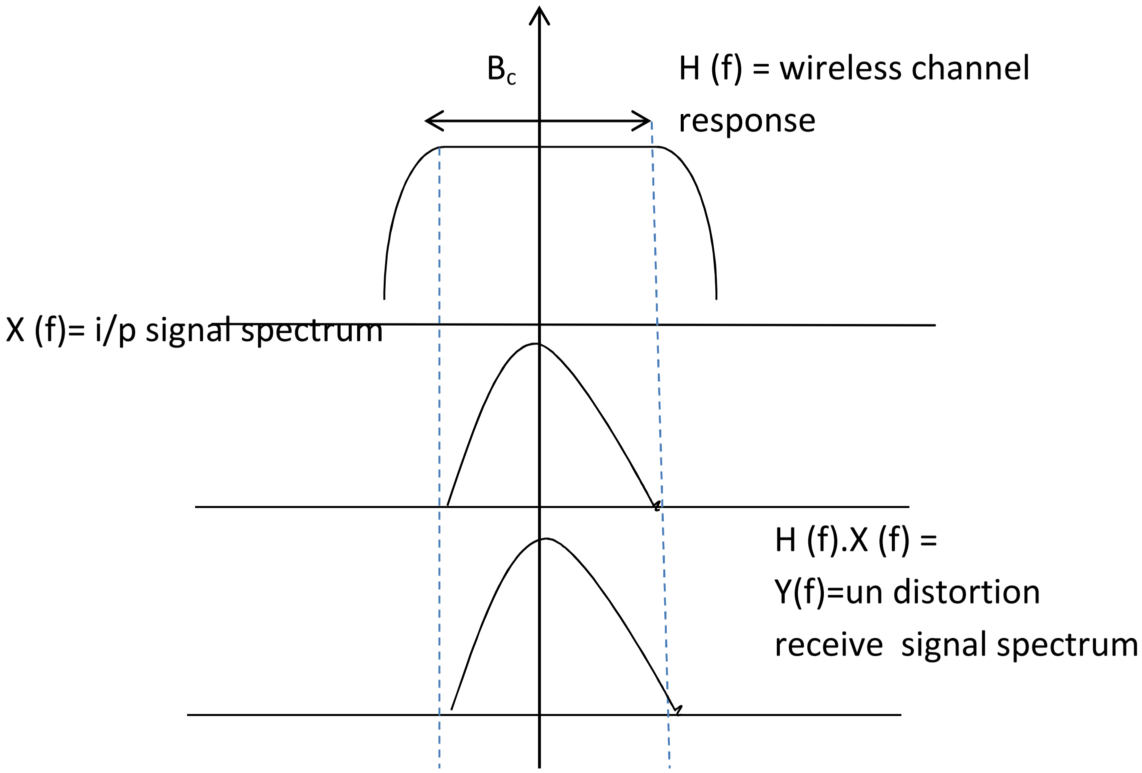


X (f) X (f).H (f)

Transmitted signal wireless channel **spectrum of the received Signal**

Let us transmit a signal with spectrum X(f) through a system which has a frequency response H(f) then the output response is simply X(f) times H(f) i.e., X(f) . H(f) is nothing but the received signal spectrum or the received spectrum.

Y(f) = X(f).H(f)



As shown in the figure, if the bandwidth Bs of the signal x(t) is less than Bc, then x(f) spans the part of the channel response H(f). Hence, the output Y(f) =H(f).X(f) is simply a scaled version of X(f) corresponding to the magnitude of flat part. Thus, input signal spectrum X(f) is undistorted at the output. Such a wireless channel is termed as a “flat - fading channel”.

$B_s \leq B_c$ = No distortion in received signal i.e. Flat fading channel.

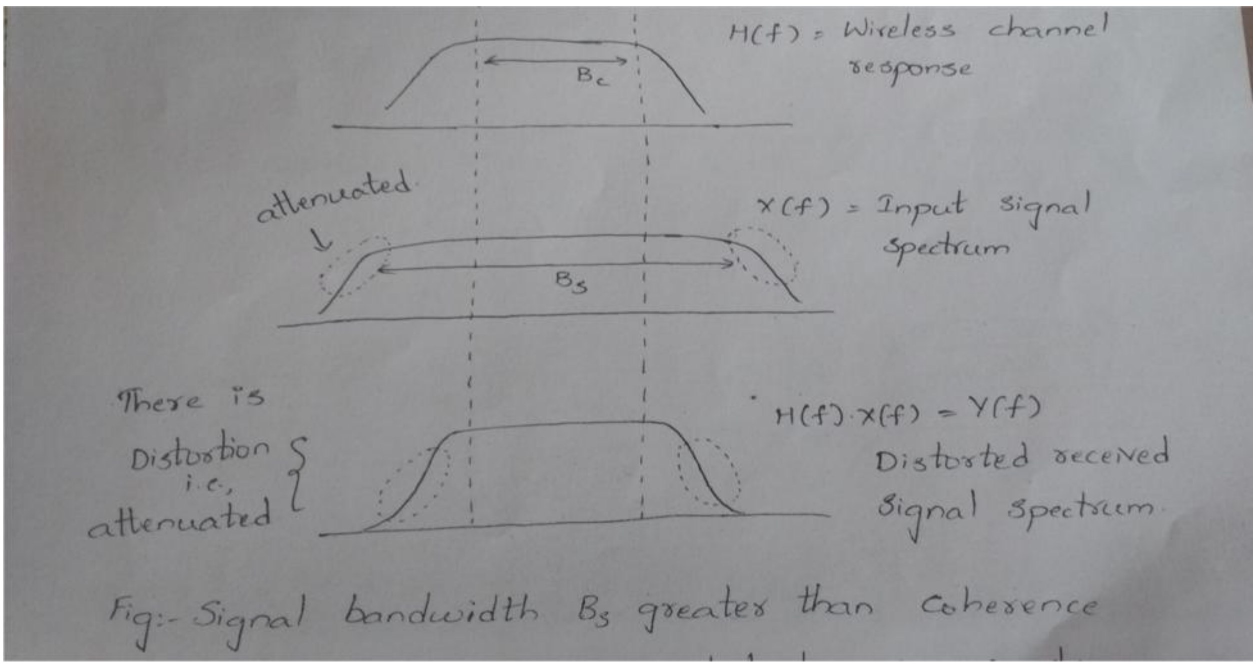
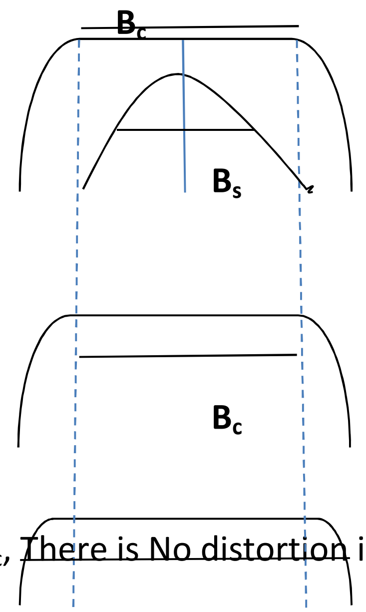


Fig: signal bandwidth B_s greater than coherence

Bandwidth B_c leading to distortion in spectrum of received signal.

Consider the case where the signal bandwidth B_s is greater than the coherence bandwidth B_c. In this scenario, different attenuations, i.e., the attenuation is frequency selective. Thus, the output spectrum Y (f) such a wireless channel is termed as a “frequency selective distortion” due to the frequency dependent nature of the attenuation of the signal.



$B_s<B_c$, No distortion flat fading ch.

$B_s>B_c$, there is distortion freq.
Frequency Selective Distortion (or) Fading

$B_s<B_c$, There is No distortion in received signal.

$B_s>B_c$, There is Distortion in received signal.

Consider a wireless delay profile H(τ) $\sum_{i=0}^{l-1} ai\ s(\tau-\tau_i)$

i.e. Multipath delay channel.

The response H (f) of this channel is given as

The response H(f) of this channel is given as

$$H(f) = \int_0^\infty h(\tau)\, e^{-j2\pi f\tau}\, d\tau$$

$$= \int_0^\infty \left[\sum_{i=0}^{L-1} a_i\, \delta(\tau-\tau_i)\right] e^{-j2\pi f\tau}\, d\tau$$

$$= \int_0^\infty \left[\sum_{i=0}^{L-1} a_i\, \delta(\tau-\tau_i)\, e^{-j2\pi f\tau}\right] d\tau$$

$$\therefore H(f) = \left[\sum_{i=0}^{L-1} a_i\, e^{-j2\pi f\tau_i}\right]$$

Fourier Transform of the multipath wireless channel.

Now consider the highest frequency harmonic corresponding to $a_i\, e^{-i2\pi f \tau i}$, i.e., with phase varying at the rate τ_{L-1} .

If $f = 0$, then $a_i e^{-j2\pi f \tau_i} = a_i e^{-j2\pi (0)\tau_i}$

$$= a_i e^{0}$$
$$= a_i (1)$$
$$= a_i$$

If $f = \frac{1}{4\tau_i}$, then $a_i e^{-j2\pi f \tau_i} = a_i e^{-j2\pi \left(\frac{1}{4\tau_i}\right)\tau_i}$

$$= a_i e^{-j\pi/2}$$
$$= a_i \left[\cos \pi/2 - j \sin \pi/2\right]$$
$$= a_i \left[0 - j(1)\right]$$

(20)

(17)

$$= a_i [0 - j]$$
$$= -ja_i$$

→ Thus, the bandwidth of the response $H(f)$ is approximately given as

$$f_c = \frac{1}{4\tau_i} \rightarrow i^{th} \text{ path}$$

$$f_c \simeq \frac{1}{4\sigma_\tau}.$$

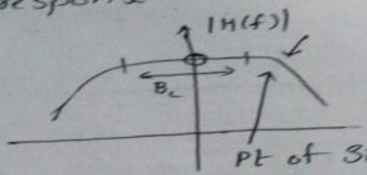


Fig:- Coherence bandwidth showing pt of change of response.

→ Hence, the coherence bandwidth of the filter $H(f)$ is approximately given as

$$B_c \approx 2f_c$$
$$= \cancel{2} \times \frac{1}{\cancel{4}_2 \sigma_\tau}$$
$$\therefore B_c = \frac{1}{2\sigma_\tau}$$

B_c = Coherence BW of the system.

σ_τ = RMS Delay spread.

$$B_c = \frac{1}{2\sigma_\tau}$$
$$= \frac{1}{2\tau_i} \quad [\because \tau_i = \sigma_\tau]$$

$$\therefore B_c \propto \frac{1}{\sigma_\tau}$$

∴ Coherence Bandwidth is inversely proportional to the RMS delay spread.

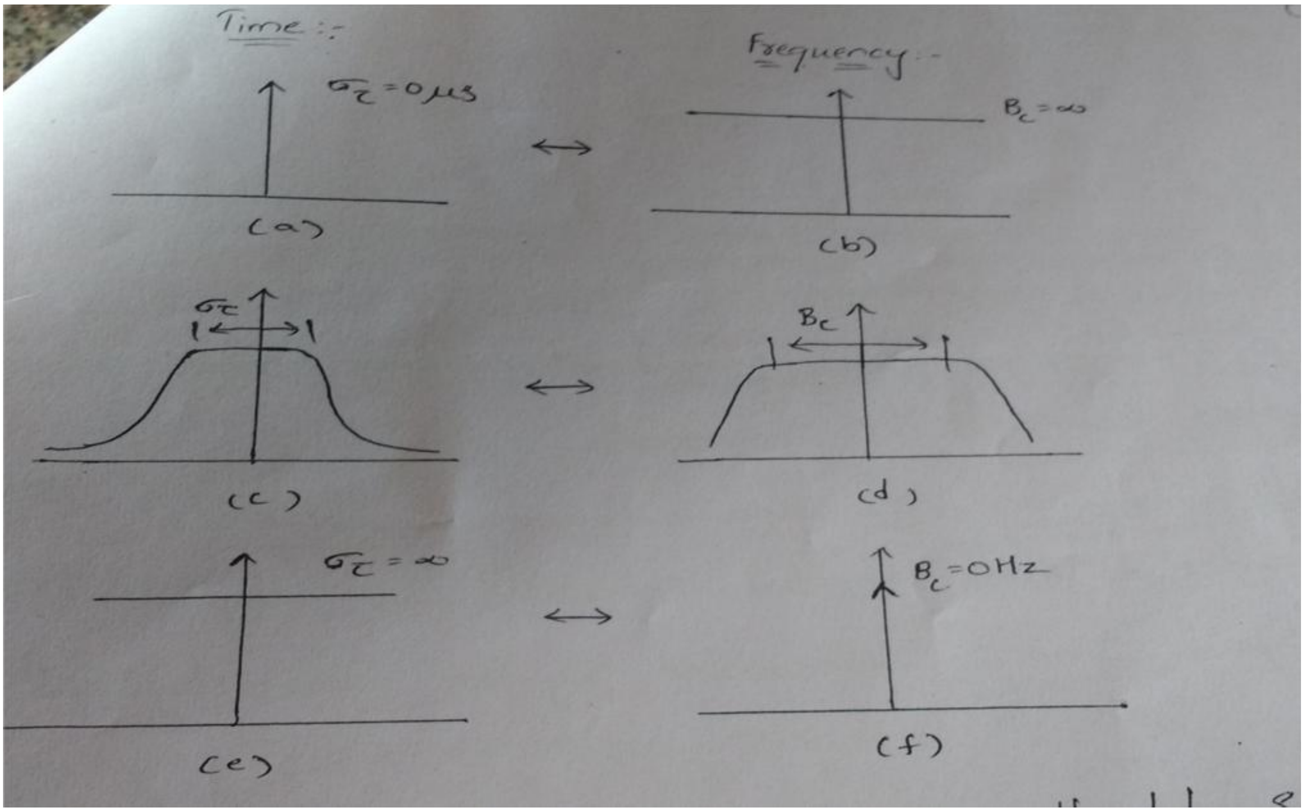


Fig: Coherence bandwidth Bc can be related to the delay spread Sigma τ as

Thus, the coherence bandwidth B_c can be related to the delay spread

Sigma τ as

$$B_c \approx 1/\sigma\tau$$

The above relation satisfies the intuitive property that coherence bandwidth $B_c$decreases as the delay spread sigma τ increases.

Finally, the approximate delay spread corresponding to outdoor channels with a typical delay spread of $2\mu s$

Can be derived as

$$B_c = 1/2\sigma\tau \quad = 1/2(2*10^{-6})$$

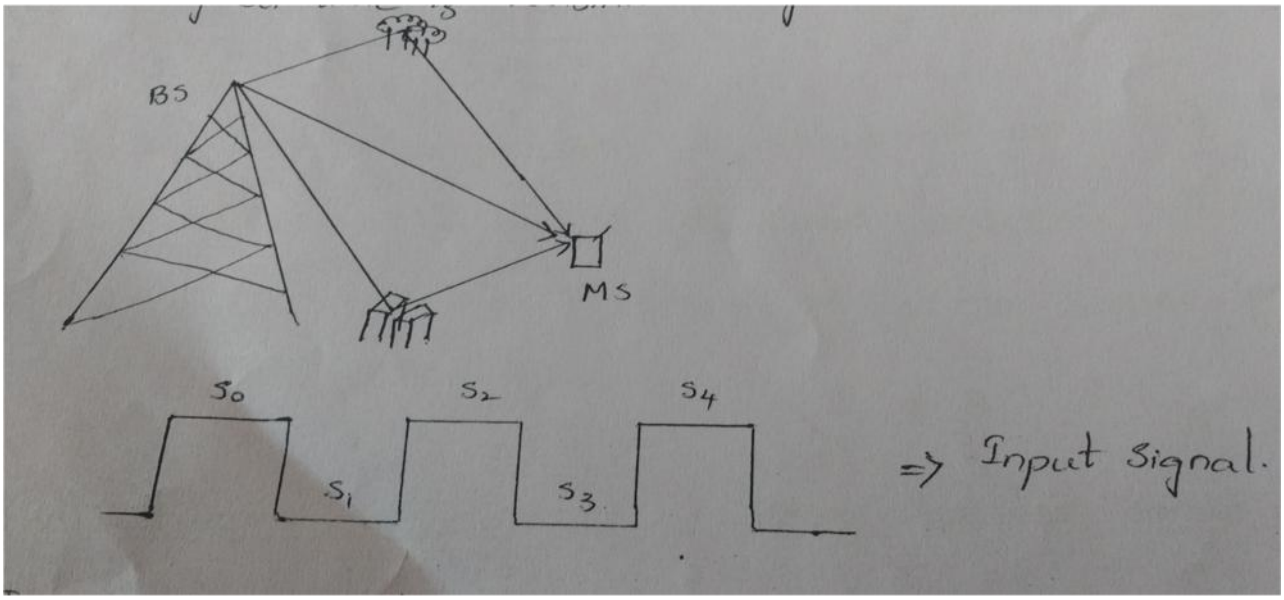
$$= 10/4^6$$

$$B_c = 250 \text{ kHz}$$

Thus, the typical delay spread of outdoor cellular wireless channels is B_c = 250

RELATION BETWEEN ISI AND COHERENCE BANDWIDTH*:*

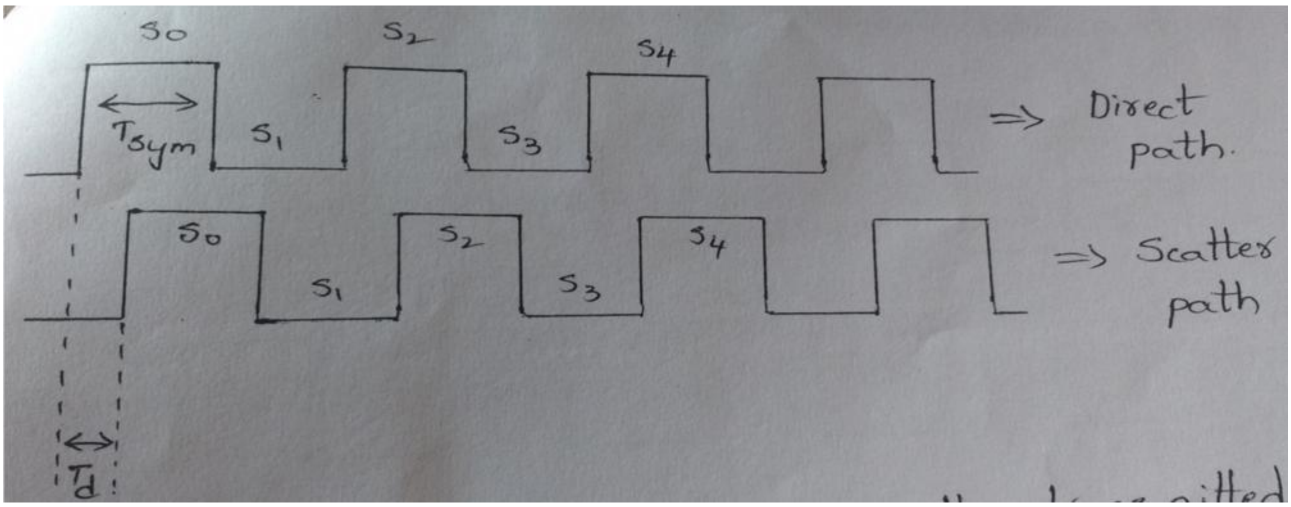
Consider a pulse amplitude modulated (PAM) signal x(t) of symbol time $T_s$transmitted by the base station.



The above signal is the transmitted signal from the base station to the mobile station.

At the receiver, the signal may be obtained from the direct loss path (LOS path) and scatter or indirect path (NLOS path).

At The Receiver:



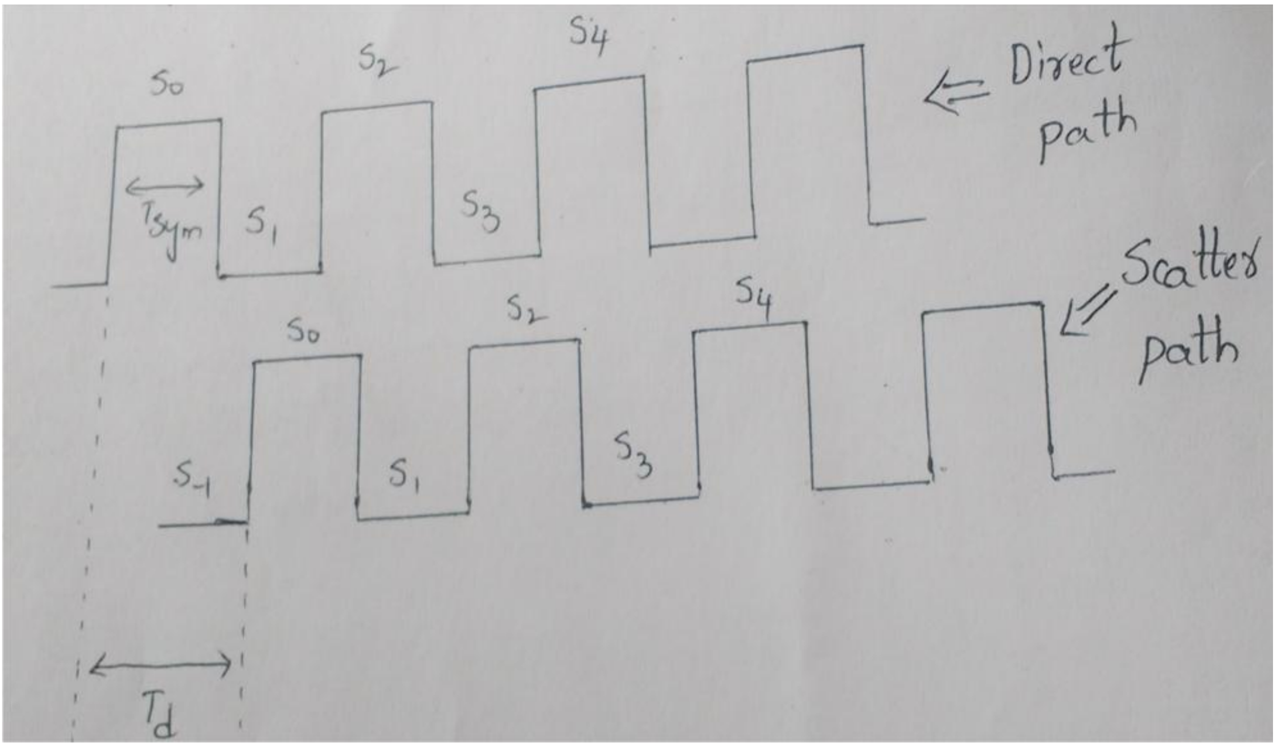
The direct path signal is same as the transmitted signal whereas the scatter path signal is obtained with some delay i.e., T_d.

$T_d = \sigma\tau$ = delay spread of the channel.

$\sigma \ll T_{sym}$ i.e., is very small.

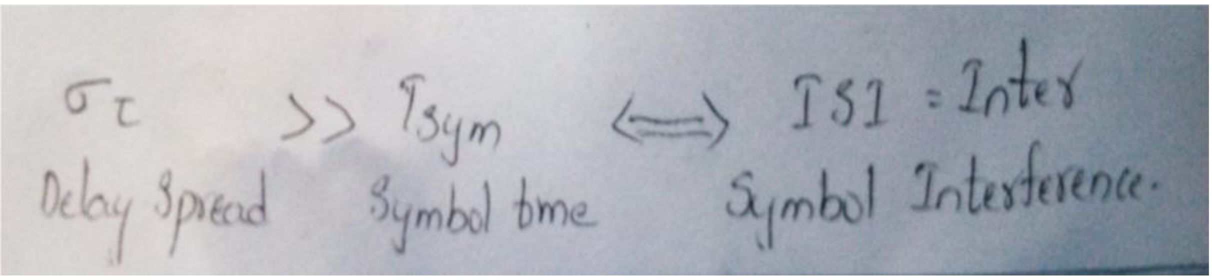
The direct path and the scattered paths components are added up to result into interference at the receiver, but here when we add these two symbols, S_0 interference with So interferes with so,S_1 interferes with S_1,S_2interferes with S_2i.e. the same symbol is interfering with each at time instant because sigma τ is very small. It is more specifically much smaller than T symbol (T_{sym}). There is no result of ISI (Inter Symbol Interference) occurrence.

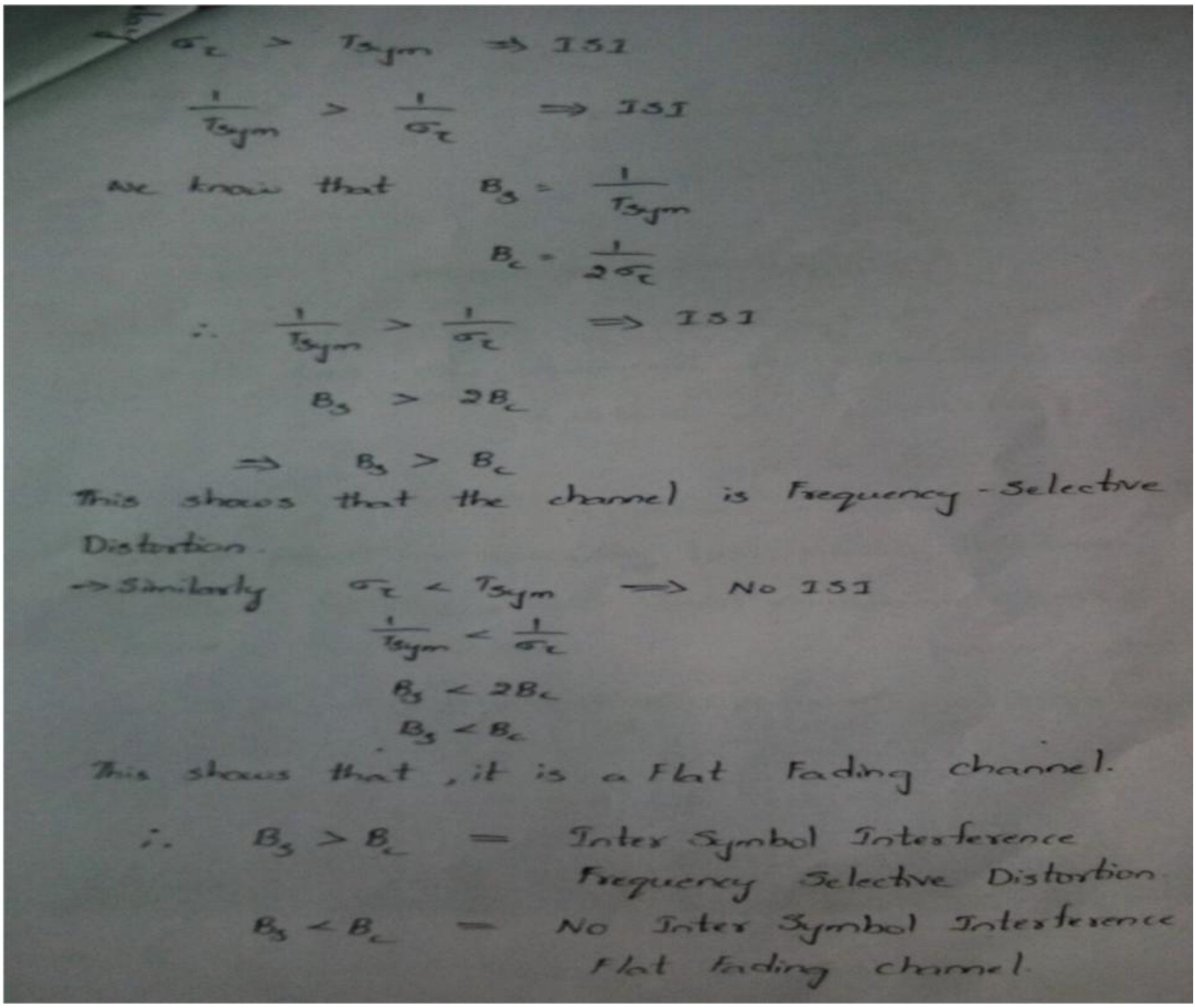
Now, let us consider a scenario in which the delay spread is much greater than symbol time.



In this case, the direct path and scattered path signals are obtained, but here the scattered path signal is having much more delay than the symbol time of the signal so we will observe S_{-1}

Now, when the signals are added up, the so interferes with S_{-1}, S_1 interferes with S_0,S_2 interferes with S_1 i.e., the previous symbol is interfering with the current symbol. Hence this results in the interference which is known as "Inter symbol Interference" (ISI).





$\sigma_\tau > T_{sym} \Rightarrow ISI$

$\frac{1}{T_{sym}} > \frac{1}{\sigma_\tau} \Rightarrow ISI$

We know that $B_s = \frac{1}{T_{sym}}$

$B_c = \frac{1}{2\sigma_\tau}$

$\therefore \frac{1}{T_{sym}} > \frac{1}{\sigma_\tau} \Rightarrow ISI$

$B_s > 2B_c$

$\Rightarrow B_s > B_c$

This shows that the channel is Frequency-Selective Distortion.

→ Similarly $\sigma_\tau < T_{sym} \Rightarrow$ No ISI

$\frac{1}{T_{sym}} < \frac{1}{\sigma_\tau}$

$B_s < 2B_c$

$B_s < B_c$

This shows that, it is a Flat Fading channel.

$\therefore$ $B_s > B_c$ — Inter Symbol Interference
Frequency Selective Distortion

$B_s < B_c$ — No Inter Symbol Interference
Flat Fading channel.

For a 3G wireless communication system, the bandwidth is considered as 5MHz i.e.,

$B_s = 5\text{MHz} = 5\times10^6 = 5000 \times 10^3$

$B_s = 5000$ kHz

$B_c = 500$ kHz.

B_s □ B_c

This condition proves that there is ISI in the channel which results into a frequency selective distortion.

Now, we have to employ some technique at the receiver, so that we can reverse this distortion in the frequency domain. The equalizer is the technique, which is employed to reverse this distortion.

Equalizer or Equalization process at the receiver is the technique used to reverse the distortion.

The delay speed is an important parameter in a 3G, 4G wireless communication systems because it characterizes that interval of time over which you are receiving copies starting from component to the scatter components, and the delay spread is inherently related to the coherence bandwidth as the display increases, the coherence bandwidth decreases and if the signal bandwidth is less than the coherence bandwidth, it does not result in frequency distortion.

If signal bandwidth is greater than the coherence bandwidth it results in Inter symbol interference in time domain i.e. frequency selective distortion. This is bad because symbols start interfering with one another making detection impossible or detection erroneous at the receiver.

<u>DOPPLER SHIFT OR FADING IN WIRELESS SYSTEMS</u>:

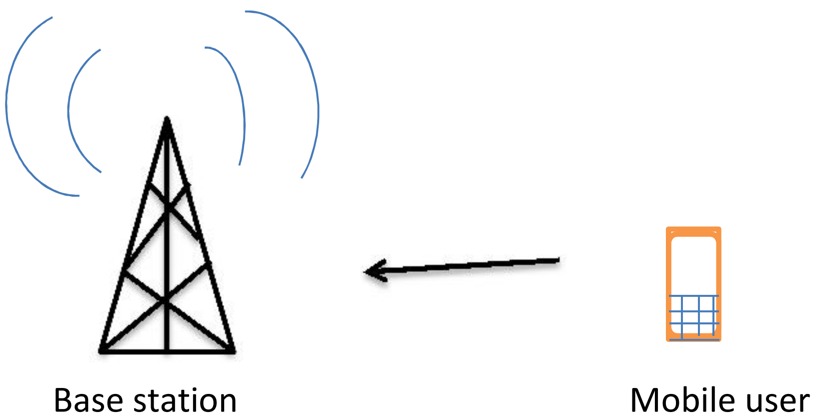


Fig: Doppler fading due to user mobility.

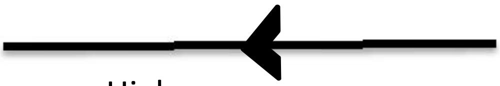
The Doppler shift is a fundamental principle related to the electromagnetic radio wave propagation. The Doppler shift is defined as the change in the frequency the wave due to relative motion between the transmitter and receiver as shown in the figure.

(Or)

Doppler shift is the frequency of the electromagnetic wave arising due to the 'relative motion' between the transmitter and the receiver.

If the receiver is moving towards the transmitter, then the received frequency is higher than the true frequency

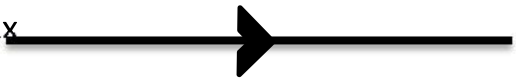
Tx Rx



Higher

Similar, if the receiver is moving away from the transmitter, then the received frequency is lower than the true frequency.

Tx Rx



Lower

So, if the motion is towards the transmitter, then the received frequency is higher and if the motion is away from the transmitter, then the received frequency is lower. Doppler shift = change in frequency of the wave due to relative motion between transmitter and receiver.

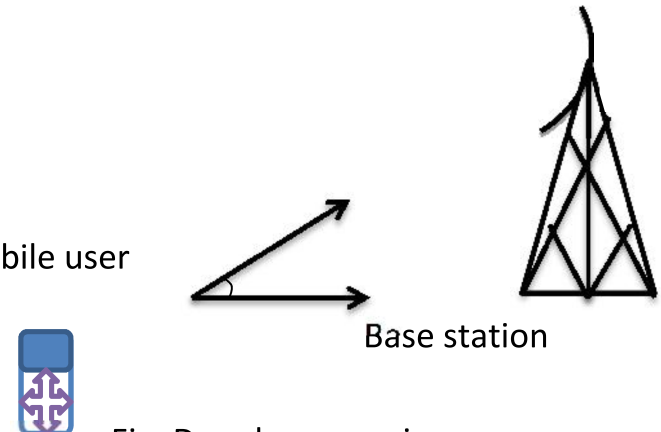


Fig: Doppler scenario

Consider the scenario shown in the figure, where the mobile station is moving with a velocity 'V' at an angle θ with the line joining the mobile and base station let the carrier frequency be Fc .The Doppler shift in the scenario is given as

$f_d = (v/c \cos\theta)\, f_c$

Where v=velocity

f_c=carrier frequency

c=Speed of light $=3*10^8$m/s

f_d=Doppler shift

The Doppler shift increases with the velocity 'V'.

Moreover, it depends critically on the angle θ between the direction of motion and the joining the transmitter and the receiver.

Example :-

1) Consider a vehicle moving at 60 miles per hour at an angle of $\theta = 0^0$ with the line joining the base station. Compute the Doppler shift of the received signal at a carrier frequency of f_c=1850MH$_z$.

Solution:- To convert the velocity from units of miles per hour to the standard meters per second ,we use

1mile =1.61 km

Given that

V=60miles per hour

=60*1.61*5/18 m/s

V=26.8 m/s.

The Doppler shift γ=f_c+(V/c cosθ)f_c

f_d= V/c cosθ f_c= 26.8/3*10^8 *cos (0^0)*1850*10^6= 165 MHz

f_d= V/c cosθ f_c= 26.8/3*10^8 *cos (0^0)*1850*10^6= 165 MHz

Thus, the Doppler shift is f_d =165Hz .since, the mobile user is moving towards the base station, Doppler shift is positive. i.e., the perceived (or) received frequency f_r is higher compared to the carrier frequency.

The received frequency is given as

F_r=f_c+f_d

= f_c+ (v cosθ/c)f_c

Where f_r= received frequency

f_c= carrier frequency

f_d = Doppler shift

The Doppler shift is maximum when$\theta = 0,\pi$ i.e. when the relative motion is along the joining the transmitter and the receiver.

$\theta = 0, \pi = f_d$ = maximum.

When $\theta = \pi/2$, i.e. the motion is perpendicular to the receive direction, Doppler shift is zero.

$$\theta = \frac{\pi}{2} = f_d = 0$$

The Doppler shift is positive in the sense, that the perceived frequency is higher if $0 \leq \theta \leq \pi/2$.

$$0 \leq \theta \leq \frac{\pi}{2} \rightarrow fd = positive, Ms \rightarrow Bs$$

On the other hand, it is negative leading a lower perceived frequency that transmit frequency is $\pi/2 \leq \theta \leq \pi$

$$\frac{\pi}{2} \leq \theta \leq \pi \rightarrow fd = negative\,; Ms \leftarrow Bs$$

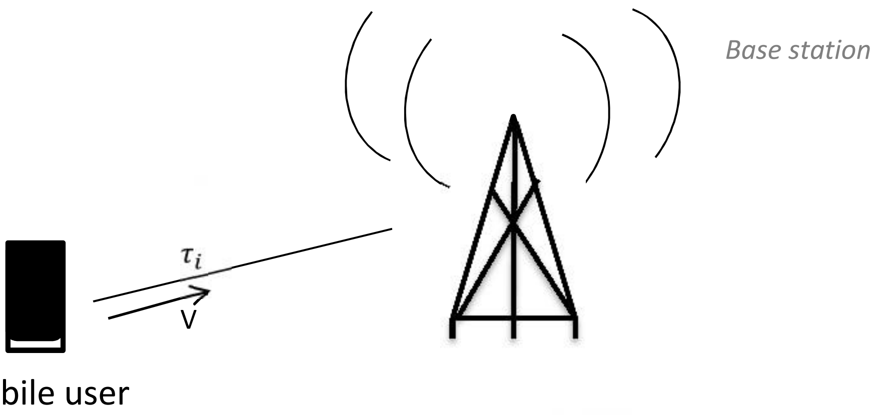
The received frequency is

$$F_r = f_c + f_d = 1850 \text{ MHz} + 165\text{Hz}$$

The Doppler shift is very small compare to the frequency of the carrier.

Doppler impact on a wireless channel:-

Baseband Channel: Channel:



- The impact of the Doppler fading is investigated on the multipath wireless-channel model. Consider the impulse response of the i^{th} component of the multipath channel given as $a_i\delta(t - \tau_i)$.

A_i= attenuation of the i^{th} path

τ_i= delay of the i^{th} path

- Let the vehicle be moving with velocity v at an angle θ with respect to the line joining the mobile and the base station as shown in figure.

- The distance between the base station and the mobile station is changing constantly due to the motion of the user.
- Therefore as a result, the delay of the i[th] signal component is also changing.

  Therefore initial propagation delay $\tau_i = \frac{di}{c}$

  Where d_i = initial distance for i^{th} signal component

  C = velocity or speed of light ($3X10^8$m/sec)

- After a small interval of time, the distance decreases by v_t

  i.e., $\tau_i - \frac{Vt}{c}$

   $\tau_i(t) = \tau_i - \frac{V\cos\theta t}{c}$, τ_i = initial delay.

  $$= \frac{di}{c} - \frac{vt\cos\theta}{c}$$

  $$= \frac{di - Vt\cos\theta}{c}$$

   $\tau_i(t) = \tau_i - \frac{Vt}{c}\cos\theta.$

- Therefore the flat-fading wireless channel coefficient is defined as.

$$h = \sum_{i=0}^{L-1} a_i e^{-j2\pi fc\tau_i(t)}$$

- The time-varying channel coefficient h is given as.

$$h(t) = \sum_{i=0}^{L-1} a_i e^{-j2\pi f_c \left(\tau_i - \frac{v\cos\theta \cdot t}{c}\right)}$$

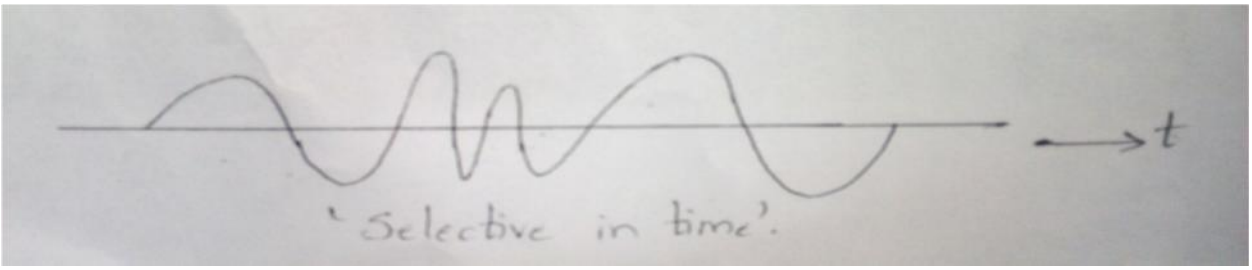
$$= \sum_{i=0}^{L-1} a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_c \frac{v\cos\theta}{c} \cdot t}$$

W.K.T $f_d = \frac{v\cos\theta}{c} \cdot t$

$$h(t) = \sum_{i=0}^{L-1} a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_d t}$$

$\therefore e^{j2\pi f_d t}$ = Time-Varying phase of wireless channel.

- The rate of variation of the phase is given by the Doppler frequency f_d.
- The mobility of the user in a wireless communication system leads to a Doppler shift, which in turn results in a time varying wireless channel coefficient. This time varying nature of wireless channel is also termed “time selectivity” and the time varying wireless channel is termed as “time selective channel”.



- So, it is selective in time, so a time varying channel also known as time selective.

Channel Coherence time of the wireless:-

- Similar to the motion of a coherence bandwidth for a frequency selective channel, now we define the concept of coherence time interval T_c for a time varying channel.
- Consider the i^{th} multipath component of the time varying channel coefficient is given as:

$$a_i(t) = a_i \, e^{-j2\pi f_c \tau_i} \, e^{j2\pi f_d t}$$

The value of this ith component corresponding to $t = 0, \pi/2$ can be obtained as

$$t=0 \Rightarrow a_i(0) = a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_d (0)}$$

$$= a_i e^{-j2\pi f_c \tau_i} e^{0}$$

$$= a_i e^{-j2\pi f_c \tau_i} (1)$$

$$a_i(0) = a_i e^{-j2\pi f_c \tau_i}$$

$$t = \frac{1}{4f_d} \Rightarrow a_i\left(\frac{1}{4f_d}\right) = a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_d \cdot \frac{1}{4f_d}}$$

$$= a_i e^{-j2\pi f_c \tau_i} e^{j\pi/2}$$

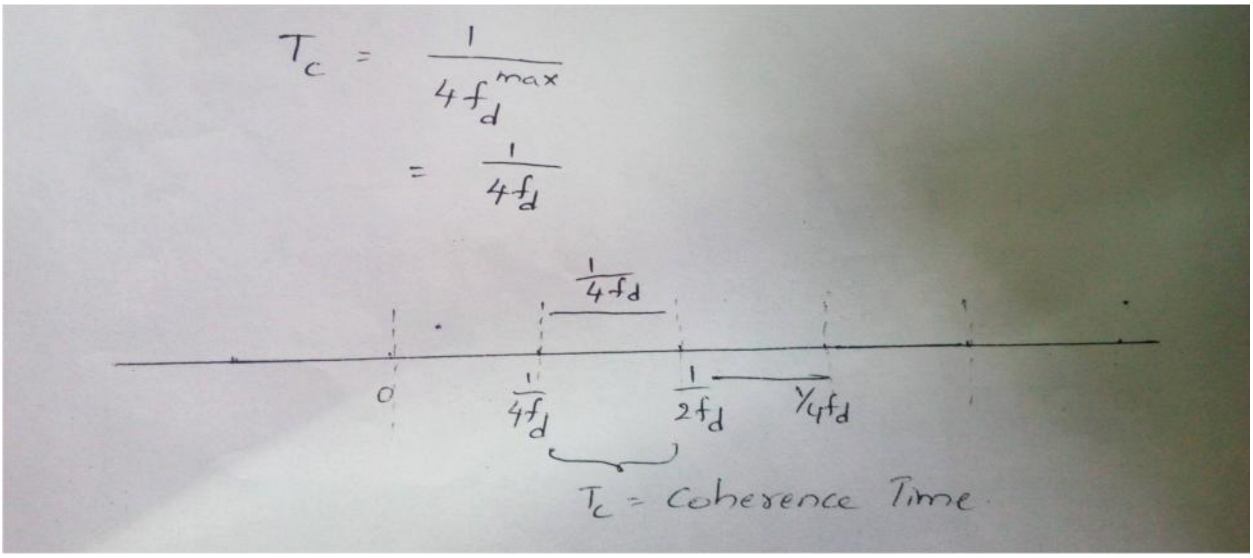
$$= a_i e^{-j2\pi f_c \tau_i} \left[\cos\pi/2 + j\sin\pi/2\right]$$

$$= a_i e^{-j2\pi f_c \tau_i} \cdot \left[0 + j(1)\right]$$

$$a_i\left(\frac{1}{4f_d}\right) = j a_i e^{-j2\pi f_c \tau_i}$$

- The channel changes significantly from time t=0 to t=1/4f_d, since the phase changes by $\frac{\pi}{2}$.

- This time duration in which the channel changes significantly due to the mobility of the user is termed as the coherence time T_c



- In the channel, if we take time 0 to $1/4f_d$, it is approximately constant in the next interval $1/4f_d$ to $1/2f_d$ which is of duration $1/f_d$, it is again approximately constant and it is changing from interval to interval and this interval of duration $1/4f_d$ is known as T_c.
- T_c= coherence time.
  = time over which channel is approximately constant.
- This can also be expressed as
  $T_c = 1/4f_d$

  $= 1/2B_d$

- Where $B_d = 2f_d$ is the Doppler spread of the wireless channel.
  Coherence time = 1/(2 X Doppler spread)
  Coherence time = T_c
  Doppler spread = B_d

Therefore $T_c = 1/2B_d$.

$B_d = 2f_d$.

- Example:
  1. consider a vehicle moving at 60miles/hour. Compute the coherence time T_c at the frequency f_c = 1.85GHz.

  So:- To compute the coherence time T_c, maximum Doppler shift f_d^{max} corresponds to

  $\theta = 0°$

Given ϑ = 60 miles/hour

1mile = 1.61km

= 60*1.61*$\frac{5}{18}$m/sec

ϑ= 26.8m/sec

$\theta = 0°$

F_c = 1.85 GHz = 1.85 * 10^9 =1850 MHz

We know that $f_d = \frac{v}{c}\cos\theta f_c$

= $\frac{26.8}{3*10^8}$ * cos(0) * 1850 * 10^6

F_d = 165Hz

 Doppler spread, $B_d = 2(f_d)^{max}$

= 2(165) = 330Hz

Coherence time, $T_c = 1/2B_d$

= 1/ 2(330) = 0.0015sec = 1.5msec

- The coherence time is approximately of the order of ms for practical wireless channel.
- For practical outdoor wireless channels the delay spread is approximately of micro sec duration ; coherence time is approximately of the order mille sec.
- Coherence bandwidth is related to the delay spread and coherence time is related to the Doppler spread
  Coherence Bandwidth B_c, σ_τ = delay spread
  Doppler spread B_d, T_c = coherence time.

Doppler spectrum:-

- Doppler spectrum gives intuition into rate of change of the wireless channel.
- This gives a clear idea of how faster or how slower is the channel varying with respect to time.

Jakes model for wireless channel correlation:

- This model gives an expression for the time correlation of the wireless channel coefficient.

- Let X,Y be two complex valued random variables. The correlation between this random variables is defined as $E\{XY^*\}$.
- A higher correlation between X, Y indicates a greater degree of similarity between the values assumed by X, Y.
- Let $a_i(t)$ denote the channel response corresponding to the i^{th} path in the multipath channel profile at time instant 't', $a_i(t)$ can be expressed as

$$a_i(t) = a_i e^{-j2\pi f c \tau_i} \underbrace{e^{j2\pi f d t}}_{\text{Time varying phase factor}}$$

- The channel coefficient $a_i(t+\Delta t)$, at time Δt later is given as.

$$a_i(t(t + \Delta t) = a_i e^{-j2\pi f c \tau_i} e^{j2\pi f d (t+\Delta t)}$$

- If the correlation between $a_i(t)$, $a_i(t+\Delta t)$ is larger, it means that $a_i(t+\Delta t)$ is vary similar to $a_i(t)$ and hence, the channel is varying very slowly.
- If the correlation between $a_i(t)$, $a_i(t+\Delta t)$ is small, it means that $a_i(t+\Delta t)$ has changed significantly compare to $a_i(t)$ and therefore, the rate of channel variation is faster.
- The time correlation coefficient; also termed the temporal correlation coefficient used to understand the rate of channel variation.
- Let this correlation as a function of Δt be denoted by $\Psi(\Delta t)$.

  $\Psi(t) = E\{a_i(t) \,.\, \underbrace{a_i^* (t+\Delta t)}_{\text{Coefficient } \Delta t \text{ later.}} \}$

$$\psi(\Delta t) = E\left\{ a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_d t} \left(a_i e^{-j2\pi f_c \tau_i} e^{j2\pi f_d (t+\Delta t)} \right)^* \right\}$$

$$= E\left\{ |a_i|^2 e^{-j2\pi f_d \Delta t} \right\}$$

$$= E\left\{ |a_i|^2 \right\} E\left\{ e^{-j2\pi f_d \Delta t} \right\}.$$

$\because E\left\{ |a_i|^2 \right\} = 1$

$\therefore \psi(\Delta t) = E\left\{ e^{-j2\pi f_d \Delta t} \right\}$

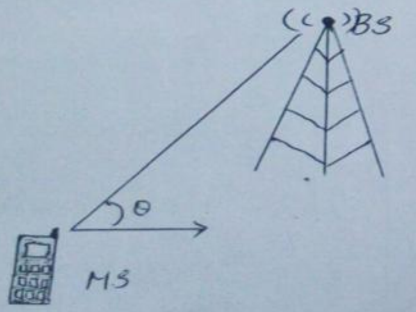
$$= E\left\{ e^{-j2\pi \left(\frac{v}{c} \cos\theta f_c\right) \Delta t} \right\}$$

Since $f_d = \frac{v}{c} \cos\theta . f_c$.

$$= E\left\{ e^{-j2\pi f_d^{max} . \cos\theta . \Delta t} \right\}$$

$$f_d^{max} = \frac{v}{c} . f_c$$

= Maximum Doppler shift.



→ Now the correlation coefficient depends on θ, which is a random quantity.

θ is uniformly distributed between $[0, \pi]$. i.e, $0 \le \theta \le \pi$. with the distribution given as $f_\theta(\theta) = \frac{1}{\pi}$.

→ Therefore, the correlation coefficient $\psi(\Delta t)$ is obtained as

$$\psi(\Delta t) = \int_0^{\pi} E\left\{ e^{-j2\pi f_d^{max} \cos\theta \, \Delta t} \right\} f_\theta(\theta)\, d\theta.$$

$$= \int_0^{\pi} E\left\{ e^{-j2\pi f_d^{max} \cos\theta \cdot \Delta t} \right\} \cdot \frac{1}{\pi}\, d\theta$$

$$= \frac{1}{\pi} \int_0^{\pi} E\left\{ e^{-j2\pi f_d^{max} \cdot \cos\theta \cdot \Delta t} \right\}$$

$$= J_0\left(2\pi f_d^{max} \cdot \Delta t\right)$$

where J_0 is the Bessel function of the 0^{th} order.

$\therefore$ Substituting $f_d^{max} = \frac{1}{4T_c}$, we get

$$\psi(\Delta t) = J_0\left(\cancel{2}\pi \cdot \frac{1}{\cancel{4}_2 T_c} \cdot \Delta t\right)$$

$$= J_0\left(\frac{\pi}{2} \cdot \frac{\Delta t}{T_c}\right).$$

→ The above temporal correlation model for $\psi(\Delta t)$ is a popular model for wireless communication systems and is termed "Jakes' model". The Jakes correlation as a function of the normalized time lag $\frac{\Delta t}{T_c}$ is shown in figure.

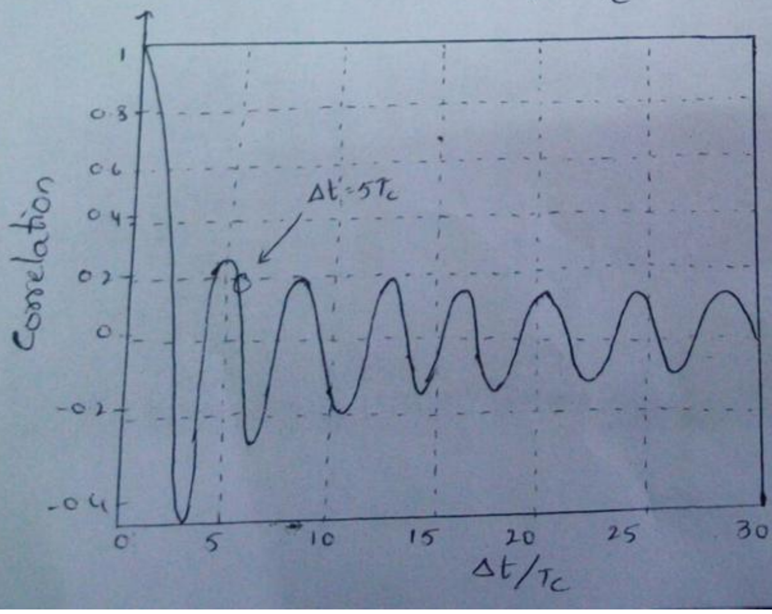


Fig:- Jakes' correlation as a function of $\frac{\Delta t}{T_c}$.

Doppler spectrum:-

- The Doppler spectrum corresponding to the temporal correlation function Ψ (Δt) is given by its Fourier transform as.

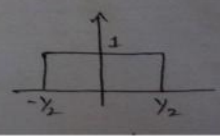
$$S_H(f) = \int_{-\infty}^{\infty} \psi(\Delta t)\, e^{-j2\pi f \Delta t}\, d(\Delta t).$$

$$= \int_{-\infty}^{\infty} J_0\left(2\pi f_d^{max} \Delta t\right) e^{-j2\pi f(\Delta t)}\, d(\Delta t).$$

$$= \frac{1}{\pi f_d^{max}} \cdot \frac{\mathrm{rect}\left(\frac{f}{2 f_d^{max}}\right)}{\sqrt{1-\left(\frac{f}{f_d^{max}}\right)^2}}.$$

where the function rect(·) is defined as

$$\mathrm{rect}(x) = \begin{cases} 1 & |x| \leq \frac{1}{2}. \\ 0 & |x| > \frac{1}{2}. \end{cases}$$



- i.e., rect (x) is basically a pulse of height 1 between x = -1/2 and x = ½. The Doppler spectrum $S_H(f)$, which is associated with the Jakes temporal correlation model, is termed. The jakes spectrum and is very popular in wireless communication to model the correlation functions of time varying wireless channels. A figure of the Jakes spectrum is shown in the figure. It can be seen that the spectrum is 'U' shaped and restricted between $-f_d{}^{max}$ and $f_d{}^{max}$. Hence this is colloquially also termed as a U- shaped Doppler spectrum of Doppler spread $2f_d{}^{max} = 2\frac{\vartheta}{c} f_c$.

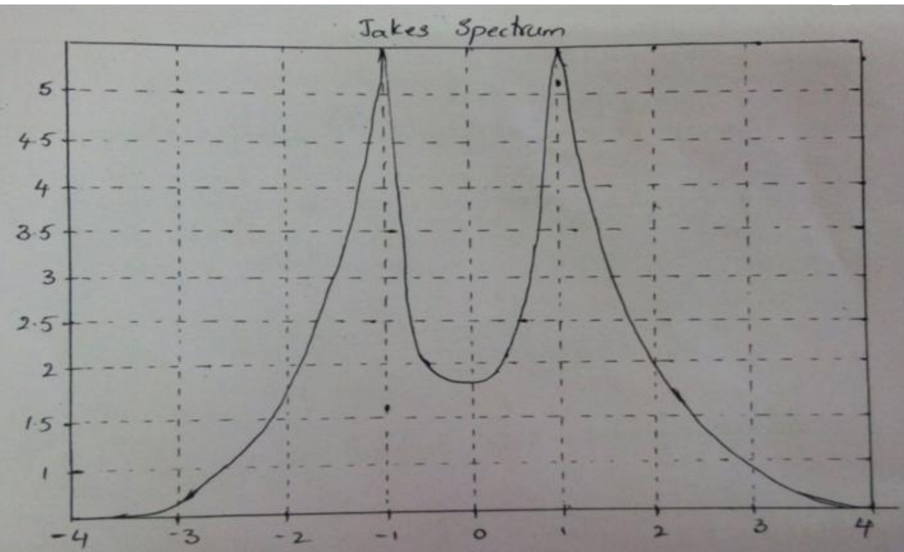
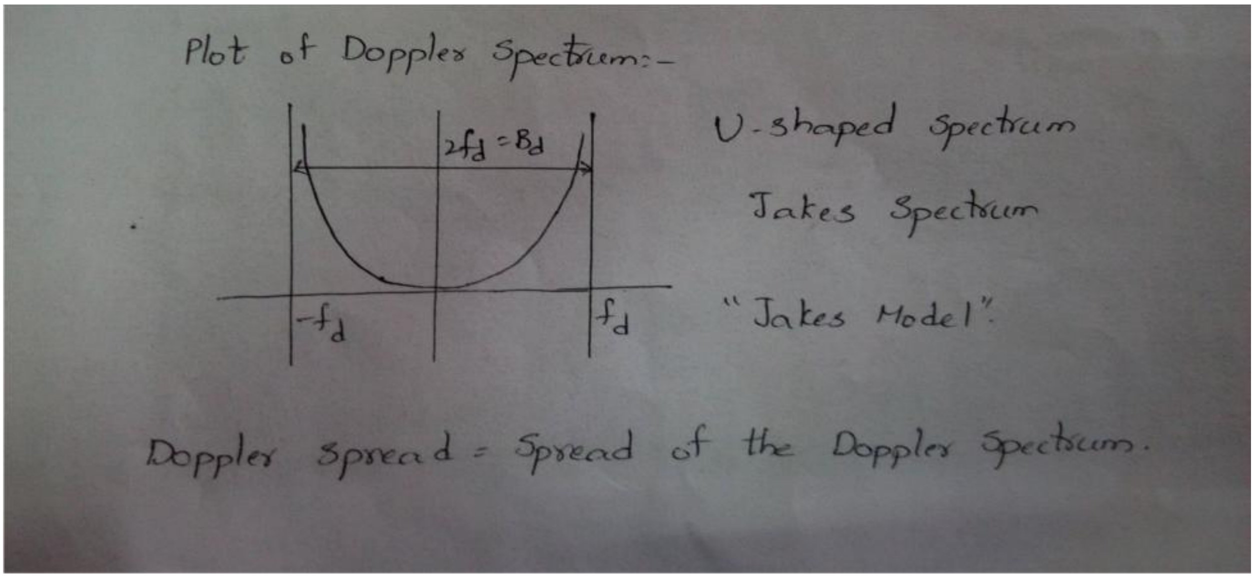


Fig:- Jakes' Spectrum as a function of normalized frequency $\frac{f}{f_d}$.



Implication of coherence time:-

- The channel is changing constant for coherence time, and channel is changing from coherence time to coherence time. Hence, to get knowledge of the channel, it needs to be 'measured' at least once every coherence time.

$$Y(t) = h\,x(t)$$

$$\hat{x}(t) = \frac{1}{h}\,y(t) = x(t)$$

Inverting channel or channel inversion

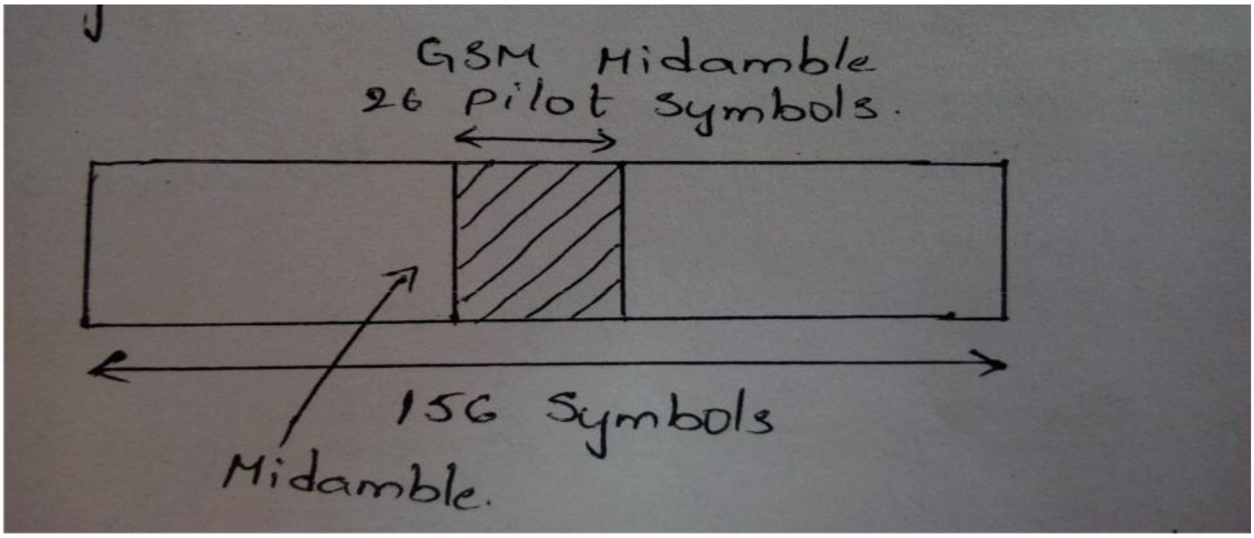
- For the detection of x(t), we need the knowledge of h. measuring or estimating 'h' is known as "CHANNEL ESTIMATION". It is a key procedure because h is needed for detecting the transmitted signal. Hence the channel estimation is a key procedure for every 3G, 4G wireless communication system.

1. How is the channel estimated at the receiver?

   By using 'Training' or 'Pilot' signals. So, the channel estimation has to be carried out the receiver we employed training, we train the receiver to give it measure estimate of the channel; these are known as trainee or pilot. Pilot are something that are transmitted in front of the actual things of the pilot symbols, or symbol that attracts transmitted in front of the information symbol, that is before transformation of the information symbols. These are employed essential to estimate the channel at the receiver.

- Example:-

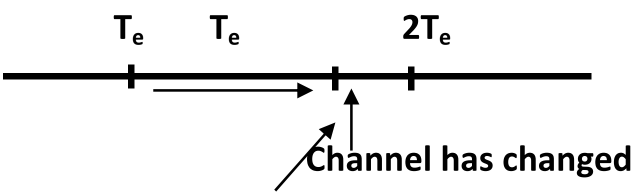
  Consider a 2G GSM, it is very popular. Every symbol every frame there is a frame for a user, for every there is a slot for every user.



- Every slot consists of 156 symbols, out of which 26 symbols are training symbol. In fact the middle of the slot, if anything is in front of something this is known as a preamble, if it is middle it is known as midamble.

Fading:-

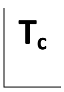
- If we coherence time 'T_c' is greater than the inter channel estimation time or the time interval between
- Two channel estimation procedures. Then the channel is not changing during one channel estimation procedure, hence this channel is a slow fading channel.
- Similarly, if the coherence time' T_c' is less than the inter channel estimation time or the interval between the two channel estimation procedures. This channel is fast fading channel.



This result in errors at receiver

- T_c > inter channel estimation time Hence 'slow fading'
- T_c< inter channel estimation time Hence 'fast fading'

- The figure below shows the relationship between key parameters of the wireless channel.



	Slow- fading **Flat- fading** **No- ISI**	**Slow – fading** **Frequency selective(ISI)**
T	**Fast – fading** **Flat – fading** **No - ISI**	**Fast – fading** **Frequency selective (ISI)**
	Ts	$\sigma\tau$

T

Fig: figure summarizing relationship between key parameters of the wireless channel.

- The plot is divided into 4 quadrants.
- If delay spread becomes greater than the symbol time so, this channel becomes frequency selective.
- If delay spread becomes lesser than the symbol time, this channel becomes flat fading.
- If coherence time is high, then the channel is varying at the slow rate, hence the channel is slow fading.
- If coherence time is low, then the channel is fast fading.

Under spread channels:-

- The delay spread of an outdoor a channel is approximately of the order of micro seconds.

$$\text{i.e., } \sigma\tau \approx \mu s$$

- The coherence time is typically of the order of mille seconds (ms).

$$T_c \approx \text{ms}$$

In typical channel,

$$\sigma\tau \ll Tc$$

- i.e., delay spread is 1000 times smaller than the coherence time.

$$\underbrace{\text{Delay spread} << \text{coherence time}}_{\text{under spread channels}}$$

UNIT-II
CELLULAR COMMUNICATION

Limitations of conventional mobile systems over cellular mobile system

Limitations of conventional mobile telephone systems: One of many reasons for developing *a* cellular mobile telephone system and deploying it in many cities is the operational limitations of conventional mobile telephone systems: limited service capability, poor service performance, and inefficient frequency spectrum utilization.

1. Limited service capability: A conventional mobile telephone system is usually designed by selecting one or more channels from a specific frequency allocation for use in autonomous geographic zones, as shown in Fig.1.1. The communications coverage area of each zone is normally planned to be as large as possible, which means that the transmitted power should be as high as the federal specification allows. The user who starts a call in one zone has to reinitiate the call when moving into a new zone because the call will be dropped. This is an undesirable radio telephone system since there is no guarantee that a call can be completed without a handoff capability. The handoff is a process of automatically changing frequencies as the mobile unit moves into a different frequency zone so that the conversation can be continued in a new frequency zone without redialing. Another disadvantage of the conventional system is that the number of active users is limited to the number of channels assigned to a particular frequency zone.

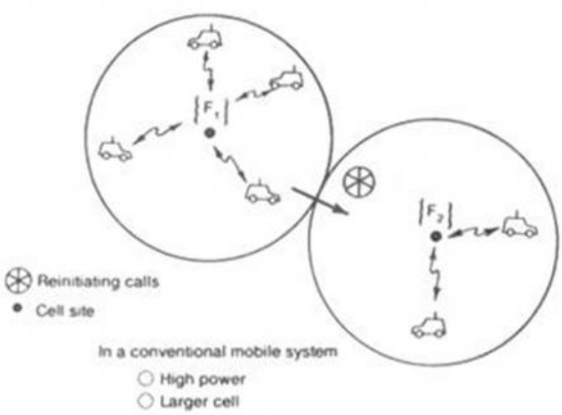


Fig1.1 Conventional Mobile System

2. Poor Service Performance: In the past, a total of 33 channels were all allocated to three mobile telephone systems: Mobile Telephone Service (MTS), Improved Mobile Telephone Service (IMTS) MJ systems, and Improved Mobile Telephone Service (IMTS) MK systems. MTS operates around 40 MHz and MJ operates at 150 MHs; both provide 11 channels; IMTS MK operates at 450 MHz and provides 12 channels. These 33 channels must cover an area 50 mi in diameter. In 1976, New York City had 6 channels of(MJ serving 320 customers, with another 2400 customers on a waiting list. New York City also had 6 channels of MK serving 225 customers, with another 1300 customers on a waiting list. The large number of subscribers created a high blocking probability during busy hours. Although service performance was undesirable, the demand was still great. A high-capacity system for mobile telephones was needed.

3. Inefficient Frequency Spectrum Utilization: In a conventional mobile telephone system, the frequency utilization measurement Mo, is defined as the maximum number of customers that could be served by one channel at the busy hour.

Mo = Number of customers/channel

Mo = 53 for MJ, 37 for MK

The offered load can then be obtained by

A = Average calling time (minutes) x total customers / 60 min (Erlangs)

Assume average calling time = 1.76 min.

A1 = 1.76 * 53 * 6 / 60 = 9.33 Erlangs (MJ system)

A2 = 1.76 * 37 * 6 / 60 = 6.51 Erlangs (MKsystem)

If the number of channels is 6 and the offered loads are A1 = 9.33 and A2 = 6.51, then from the Erlang B model the blocking probabilities, B1 = 50 percent (MJ system) and B2 =30 percent (MK system), respectively. It is likely that half the initiating calls will be blocked in the MJ system, a very high blocking probability. As far as frequency spectrum utilization is concerned, the conventional system does not utilize the spectrum efficiently since each channel can only serve one customer at a time in a whole area. This is overcome by the new cellular system.

Truncking Efficiency:

To explore the trucking efficiency degradation inherent in licensing two or more carriers rather than one, compare the truncking efficiency between one cellular system per market operating 666 channels and two cellular systems per market each operating 333 channels. Assume that all frequency channels are evenly divided into seven subareas called cells. In each cell, the blocking probability of 0.02 is assumed. Also the average calling time is assumed to be 1.76 min.

With N1=666/7 = 95 and B= 0.02 to obtain the offered load A1 =83.1 and with N2=333/7=47.5 and B=0.02 to obtain A2= 38. Since two carriers each operating 333 channels are considered, the total offered load is 2A. We then realize that $A_1 \geq 2A_2$

converting above eqn. to the number of users who can be served in a busy hour, the average calling time of 1.76 mm is introduced. The number of calls per hour served in a cell can be expressed as

$$Q_i = \frac{A \times 60}{1.76} \text{ calls/h}$$

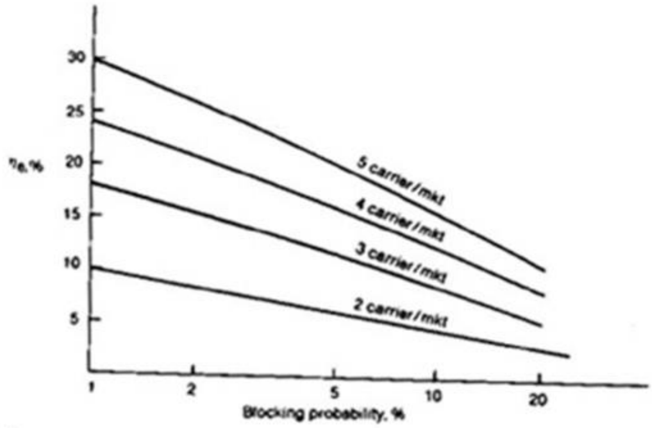


Fig. 1.2: Degradation of truncking efficiency

$$Q_i = \begin{cases} 2832.95 \text{ calls/h} & \text{(1 carrier/market)} \\ 1295.45 \times 2 = 2590.9 \text{ calls/h} & \text{(2 carriers/market)} \end{cases}$$

The truncking efficiency factor can be calculated as

$$\eta_e = \frac{2832.95 - 2590.9}{2832.95} = 8.5\%$$

For a blocking probability of 2 percent, Figure 13 shows, by comparing one carrier per market with more than one carrier per market situations with different blocking Probability conditions. The degradation of trunking efficiency decreases as the blocking probability increases. As the number of carriers per market increases the degradation increases. However, when a high percentage of blocking probability, say more than 20 percent, occurs, the performance of one carrier per market is already so poor that further degradation becomes insignificant, as Fig.1.2 shows. For a 2 percent blocking probability, the truncking efficiency of one carrier per market does show a greater advantage when compared to other scenarios.

Performance Criteria

- Voice Quality
- Service Quality
- Special Features

1. Voice Quality

A set value *x* at which *y* percent of customers rate the system voice quality (from transmitter to receiver) as good or excellent, the top two circuit merits (CM) of the five listed below.

CM	Score	Quality scale
CM5	5	Excellent (speech perfectly understandable)
CM4	4	Good (speech easily understandable, some noise)
CM3	3	Fair (speech understandable with a slight effort, occasional repetitions needed)
CM2	2	Poor (speech understandable only with considerable effort, frequent repetitions needed)
CM1	1	Unsatisfactory (speech not understandable)

2. Service Quality

- Coverage

The system should serve an area as large as possible. The transmitted power would have to be very high to illuminate weak spots with sufficient reception, a significant added cost factor. The higher the transmitted power, the harder it becomes to control interference.

- Required grade of service

The grade of service specified for a blocking probability of 0.02 for initiating calls at the busy hour. This is an average value. However, the blocking probability at each cell site will be different. To decrease the blocking probability requires a good system plan and a sufficient number of radio channels.

- Number of dropped calls

A high drop rate could be caused by either coverage problems or handoff problems related to inadequate channel availability.

3. Special Features

- call forwarding
- call waiting
- voice stored (VSR) box

- automatic roaming, or navigation services

All these services are expected at no extra cost.

Basic Cellular System

A basic cellular system consists of three parts: a mobile unit, a cell site, and a mobile telephone switching office (MTSO), as Fig 1.6 shows, with connections to link the three sub systems.

1.Mobile units: A mobile telephone unit contains a control unit, a transceiver, and an antenna system.

2.Cell site: The cell site provides interface between the MTSO and the mobile units. it has a control unit, radio cabinets, antennas, a power plant, and data terminals.

3.MTSO: The switching office, the central coordinating element for all cell sites, contains the cellular processor and cellular switch. It interfaces with telephone company zone offices, controls call processing, and handles billing activities.

4. Connections: The radio and high-speed data links connect the three subsystems. Each mobile unit can only use one channel at a time for its communication link. But the channel is not fixed: it can be any one in the entire band assigned by the serving area, with each site having multichannel capabilities that can connect simultaneously to many mobile units.

The MTSO is the heart of the cellular mobile system. Its processor provides central coordination and cellular administration. The cellular switch, which can be either analog or digital, switches calls to connect mobile subscribers to other mobile subscribers and to the nationwide telephone network. It uses voice trunks similar to telephone company interoffice voice trunks. It also contains data links providing supervision links between the processor and the switch and between the cell sites and the processor. The radio link carries the voice and signaling between the mobile unit and the cell site. The high-speed data links cannot be transmitted over the standard telephone trunks and therefore must use either microwave links or T-carriers (wire lines). Microwave radio links or T-carriers carry both voice and data between the cell site and the MTSO.

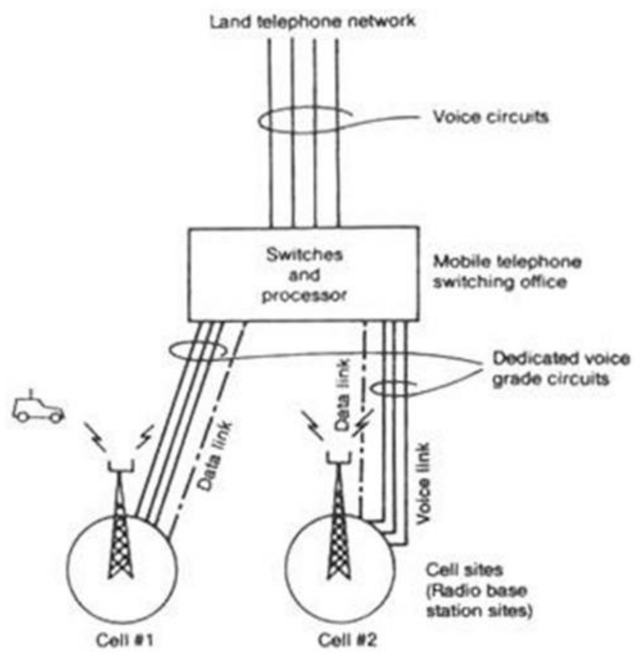


Fig.1.6: Basic cellular system

Hexagonal-shaped cells: The hexagonal-shaped communication cells are artificial and that such a shape cannot be generated in the real world. Engineers draw hexagonal-shaped cell on a layout to simplify the planning and design of a cellular system because it approaches a circular shape that is the ideal power coverage area. The circular shapes have overlapped areas which make the drawing unclear. The hexagonal-shaped cells fit the planned area nicely, as shown in Fig.1.7 with no gap and no overlap between the hexagonal cells. The ideal cell shapes as well as the real cell shapes are also shown in Fig.1.7. A simple mechanism which makes the cellular system implement- able based on hexagonal cells will be illustrated in later chapters. Otherwise, a statistical approach will be used in dealing with a real-world situation. Fortunately, the outcomes resulting from these two approaches are very close, yet the latter does not provide a clear physical picture, as shown later. Besides, today these hexagonal-shaped cells have already become a widely promoted symbol for cellular mobile systems.

An analysis using hexagonal cells, if it is desired, can easily be adapted by the reader.

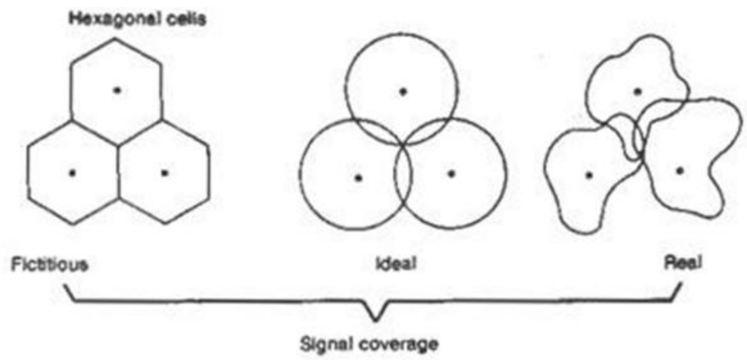


Fig.1.7 Hexagonal Cells and the real shapes of their coverage

General view of telecommunication and the function of the each unit

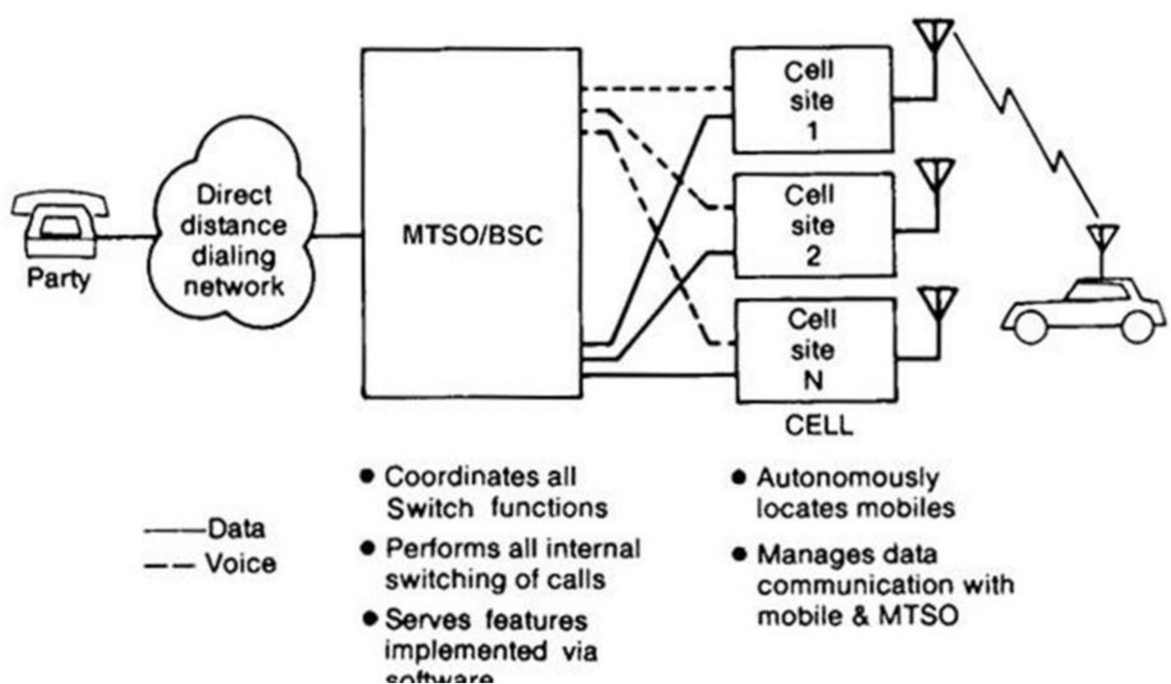


Fig.1.8: A general view of cellular telecommunications system

Antenna:

Antenna pattern, antenna gain, antenna tilting, and antenna height 6 all affect the cellular system design. The antenna pattern can be omnidirectional, directional, or any shape in both the vertical and the horizon planes. Antenna gain compensates for the transmitted power. Different antenna patterns and antenna gains at the cell site and at the mobile units would affect the system performance and so must be considered in the

system design. The antenna patterns seen in cellular systems are different from the patterns seen in free space. If a mobile unit travels around a cell site in areas with many buildings, the Omni directional antenna will not duplicate the Omni pattern. In addition, if the front-to-back ratio of a directional antenna is found to be 20 dB in free space, it will be only 10 dB at the cell site. Antenna tilting can reduce the interference to the neighboring cells and enhance the weak spots in the cell. Also, the height of the cell-site antenna can affect the area and shape of the coverage in the system.

Switching Equipment:

The capacity of switching equipment in cellular systems is not based on the number of switch ports but on the capacity of the processor associated with the switches. In a big cellular system, this processor should be large. Also, because cellular systems are unlike other systems, it is important to consider when the switching equipment would reach the maximum capacity. The service life of the switching equipment is not determined by the life cycle of the equipment but by how long it takes to reach its full capacity. If the switching equipment is designed in modules, or as distributed switches, more modules can be added to increase the capacity of the equipment. For decentralized systems, digital switches may be more suitable. The future trend seems to be the utilization of system handoff. This means that switching equipment can link to other switching equipment so that a call can be carried from one system to another system without the call being dropped.

Data Links:

The data links are shown in Fig 1.8. Although they are not directly affected by the cellular system, they are important in the system. Each data link can carry multiple channel data (10 kbps data transmitted per channel) from the cell site to the MTSO. This fast-speed data transmission cannot be passed through a regular telephone line. Therefore, data bank devices are needed. They can be multiplexed, many-data channels passing through a wideband T-carrier wire line or going through a microwave radio link where the frequency is much higher than 850MHz. Leasing T1-carrier wire lines through telephone companies can be costly. Although the use of microwaves may be a long-term money saver, the availability of the microwave link has to be considered.

Digital Cellular System:

Digital cellular systems are the cellular systems that use the digital communication techniques like in modulation, transmission format and demodulation and so on. The characteristics of these systems are

1.These offer an effective data transmission compared to the conventional analog cellular systems. These systems employ the packet switched communication technique which is faster than the circuit switching technique.

2. These systems employ powerful error detection and Correction techniques, which can country the debilitating effect of noise, fading and interference on the signal.

3. These systems also provide the security on transmitting data through encryption and decryption techniques authentication.

4.These systems also require very less transmit power, this properly increases the battery life (in portable mobile units).

5.The range of services provided by the digital cellular system is quite large compared to that provided by the analog cellular systems.

6.The speed of services provided by digital systems is quite high and thus, they support high capacity data transfers.

7.The digital cellular systems employ TDMA technique for communication.

Some examples of the digital cellular systems are:

(i) GSM

(ii) NA-TDMA (North American TDMA)

(iii) CDMA

(iv) PDC

(v) 1800-DCS.

In 1992, the first digital cellular system, GSM was developed in Germany. GSM is a European standard system. In the United States, an NA-TDMA system and a CDMA system have been developed. A Japanese system, PDC (Personal Digital Cellular) was deployed in Osaka in June 1994.

UNIT II

ELEMENTS OF CELLULAR RADIO SYSTEM DESIGN

General Description of the Problem

Based on the concept of efficient spectrum utilization, Elements of Cellular Mobile Radio System Design are

(1) the concept of frequency reuse channels

(2) the co-channel interference reduction factor

(3) the desired carrier-to-interference ratio

(4) the handoff mechanism

(5) cell splitting.

Challenge - to serve the greatest number of customers with a specified system quality.

1. How many customers can be served in a busy hour?
2. How many subscribers can we take into the system?
3. How many frequency channels are needed?

Maximum Number of Calls per Hour per Cell

To calculate the predicted number of calls per hour per cell Q in each cell, the following parameters are required

- The size of the cell
- The traffic conditions in the cell

Maximum number of frequency channels per cell

The maximum number of frequency channels per cell N is closely related to an average calling time in the system. The standard user's calling habits may change as a result of the charging rate of the system and the general income profile of the users If an average calling time T is 1.76 min and the maximum calls per hour per cell is Qi, then the offered load can be derived as

$$A = Q_i * T/60 \text{ (Erlangs)}$$

If the blocking probability is given, then it is easy to find the required number of radios in each cell. If a large area is covered by 28 cells, Kt = 28; the total number of customers in the system increases. Therefore, we may assume that

$$M_i = f(Q_i, \eta_c)$$

the number of subscribers per cell Mi is somehow related to the percentage of car phones used in the busy hours and the number of calls per hour per cell Qi as

Where the value Qi is a function of the blocking probability B, the average calling time T, and the number of channels N.

$$Qi = f(B, T, N)$$

Concept of Frequency Reuse Channels: A radio channel consists of a pair of frequencies one for each direction of transmission that is used for full-duplex operation. Particular radio channels, say F1, used in one geographic zone to call a cell, say C1, with a coverage radius R can be used in another cell with the same coverage radius at a distance D away. Frequency reuse is the core concept of the cellular mobile radio system. In this frequency reuse system users in different geographic locations (different cells) may simultaneously use the same frequency channel (see Fig.2.1). The frequency reuse system can drastically increase the spectrum efficiency, but if the system is not properly designed, serious interference may occur. Interference due to the common use of the same channel is called co-channel interference and is our major concern in the concept of frequency reuse.

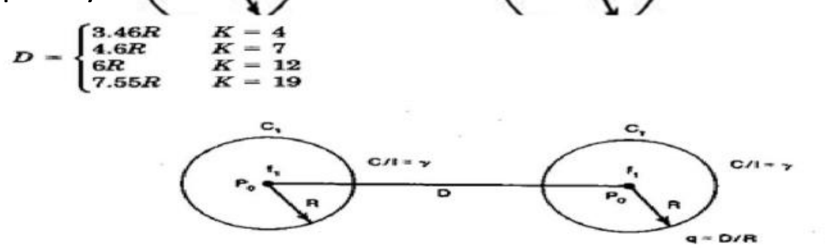


Fig.2.1 The ratio of D/R

Frequency reuse scheme: The frequency reuse concept can be used in the time domain and the space domain. Frequency reuse in the time domain results in the occupation of the same frequency in different time slots. It is called time division multiplexing (TDM). Frequency reuse in the space domain can be divided into two categories.

1.Same frequency assigned in two different geographic areas, such as A.M or FM radio stations using the same frequency in different cities.

2.Same frequency repeatedly used in a same general area in one system - the scheme is used in cellular systems. There are many co-channel cells in the system. The total frequency spectrum allocation is divided into K frequency reuse patterns, as illustrated in Fig. 2.2 for K — 4, 7, 12, and 19.

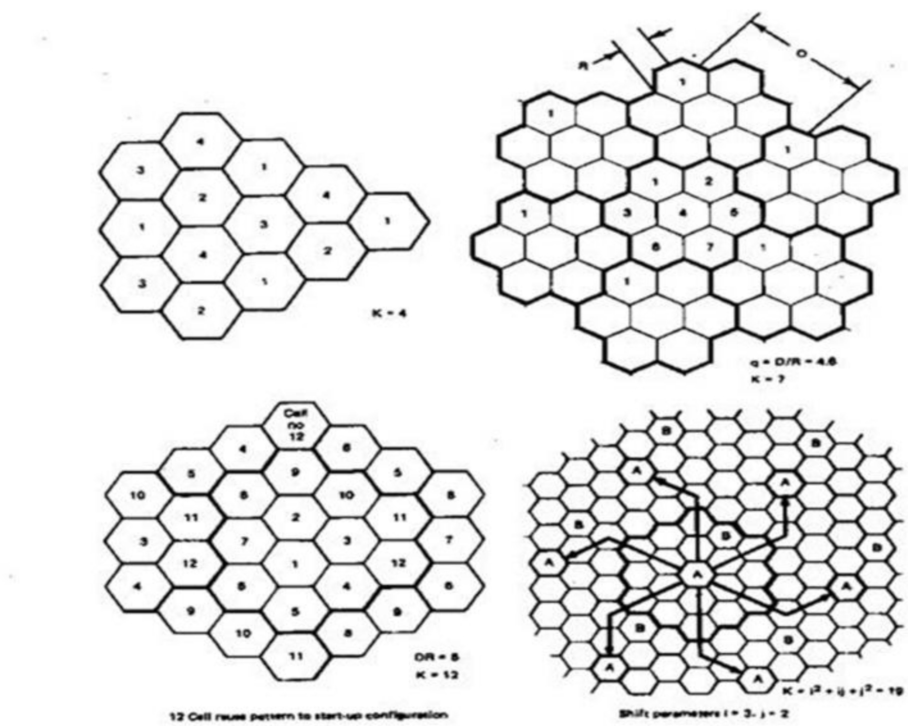


Fig.2.2 N- cell reuse pattern

Frequency reuse distance: The minimum distance which allows the same frequency to be reused will depend on many factors, such as the number of co-channel cells in the vicinity of the center cell, the type of geographical terrain contour, the antenna height and the transmitted power at each cell site. The frequency reuse distance can be determined from

Where K is the frequency reuse pattern shown in Fig.2.1, then

$$D = \begin{cases} 3.46R & K = 4 \\ 4.6R & K = 7 \\ 6R & K = 12 \\ 7.55R & K = 19 \end{cases}$$

If all the cell sites transmit the same power, then K increases and the frequency reuse distance D increases. This increased D reduces the chance that co channel interference may occur.

Theoretically, a large K is desired. However, the total number of allocated channels is fixed. When K is too large, the number of channels assigned to each of K cells becomes small. It is always true that if the total number of channels in K cells is divided as K increases, trunking inefficiency results. The same principle applies to spectrum inefficiency: if the total numbers of channels are divided into two network systems serving in the same area, spectrum inefficiency increases.

Obtaining the smallest number K involves estimating co channel interference and selecting the minimum frequency reuse distance D to reduce co channel interference. The smallest value of K is K = 3, obtained by setting i = 1, j = 1 in the equation.

Co channel interference reduction factor:

Reusing an identical frequency channel in different cells is limited by co channel interference between cells, and the co channel interference can become a major problem.

Assume that the size of all cells is roughly the same. The cell size is determined by the coverage area of the signal strength in each cell. As long as the cell size is fixed, co channel interference is independent of the transmitted power of each cell. It means that the received threshold level at the mobile unit is adjusted to the size of the cell. Actually, co channel interference is a function of a parameter q defined as

$$q = D/R$$

The parameter q is the co channel interference reduction factor. When the ratio q increases, co channel interference decreases. Furthermore, the separation D is a function of K, and C/I,

$$D=f(K,C/I)$$

Where K, is the number of co channel interfering cells in the first tier and C/I is the received carrier-to-interference ratio at the desired mobile receiver.

$$\frac{C}{I} = \frac{C}{\sum_{k=1}^{K_I} I_k}$$

In a fully equipped hexagonal-shaped cellular system, there are always six co channel interfering cells in the fist tier, as shown in Fig.2.3 ; that is, K = 6. The maximum number of K, in the first tier can be shown as six. Co channel interference can be experienced both at the cell site and at mobile units in the center cell. If the interference is much greater, then the carrier-to-interference ratio C/I at the mobile units caused by the six interfering sites is (on the average) the same as the C/I received at the center cell site caused by interfering mobile units in the six cells.

According to both the reciprocity theorem and the statistical summation of radio propagation, the two C/I values can be very close. Assume that the local noise is much less than the interference level and can be neglected. C/I then can be expressed as

$$\frac{C}{I} = \frac{R^{-\gamma}}{\sum_{k=1}^{K_I} D_k^{-\gamma}}$$

Where γ is a propagation path-loss slope determined by the actual terrain environment. In a mobile radio medium, γ usually is assumed to be 4. K is the number of co channel interfering cells and is equal to 6 in a fully developed system, as shown in Fig. 5.

The six co channel interfering cells in the second tier cause weaker interference than those in the first tier. Therefore, the co channel interference from the second tier of interfering cells is negligible

$$\frac{C}{I} = \frac{1}{\sum_{k=1}^{K_I} \left(\frac{D_k}{R}\right)^{-\gamma}} = \frac{1}{\sum_{k=1}^{K_I} (q_k)^{-\gamma}}$$

Where *qk* is the co channel interference reduction factor with *Kth* co channel interfering cell

$$q_k = \frac{D_k}{R}$$

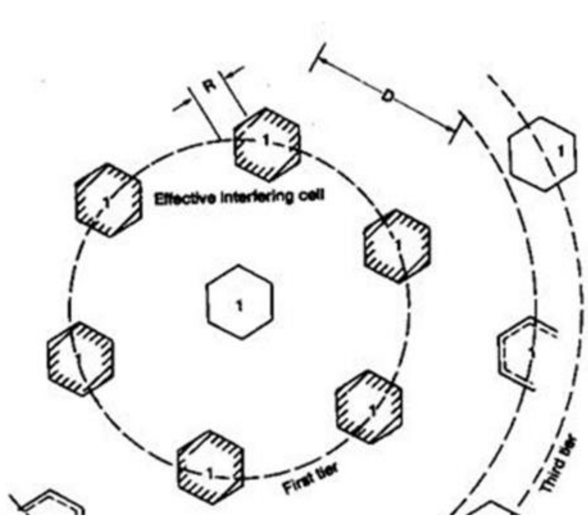


Fig 2.3: Six effective interfering cells of cell 1

C/I for normal case in an omnidirectional antenna system:

There are two cases to be considered: (1) the signal and co channel interference received by the mobile unit and (2) the signal and co channel interference received by the cell site. Both cases are shown in Fig.2.4. Nm and Nb are the local noises at the mobile unit and the cell site, respectively. Usually Nm and Nb are small and can be neglected as compared with the interference level.

As long as the received carrier-to-interference ratios at both the mobile unit and the cell site are the same, the system is called a balanced system. In a balanced system, we can choose either one of the two cases to analyze the system requirement; the results from one case are the same for the others.

Assume that all Dk are the same for simplicity, then D = Dk and q = qk,

$$\frac{C}{I} = \frac{R^{-\gamma}}{6D^{-\gamma}} = \frac{q^{\gamma}}{6}$$

Thus

$$q^{\gamma} = 6\frac{C}{I}$$

And

$$q = \left(6\frac{C}{I}\right)^{1/\gamma}$$

The value of C/I is based on the required system performance and the specified value of is based on the terrain environment. With given values of C/I and γ , the co channel interference reduction factor q can be determined. Normal cellular practice is to specify C/I to be 18 dB or higher based on subjective tests. Since a C/I of 18 dB is measured by the acceptance of voice quality from present cellular mobile receivers, this acceptance implies that both mobile radio multipath fading and co channel interference become ineffective at that level. The path-loss slope is equal to about 4 in a mobile radio environment.

$$q = D/R = (6 \times 63.1)^{1/4} = \underline{4.41}$$

The 90th percentile of the total covered area would be achieved by increasing the transmitted power at each cell; increasing the same amount of transmitted power in each cell does not affect the result. This is because q is not a function of transmitted power. The factor q can be related to the finite set of cells K in a hexagonal-shaped cellular system by

Substituting q = 4.41 in above equation yields k=7.

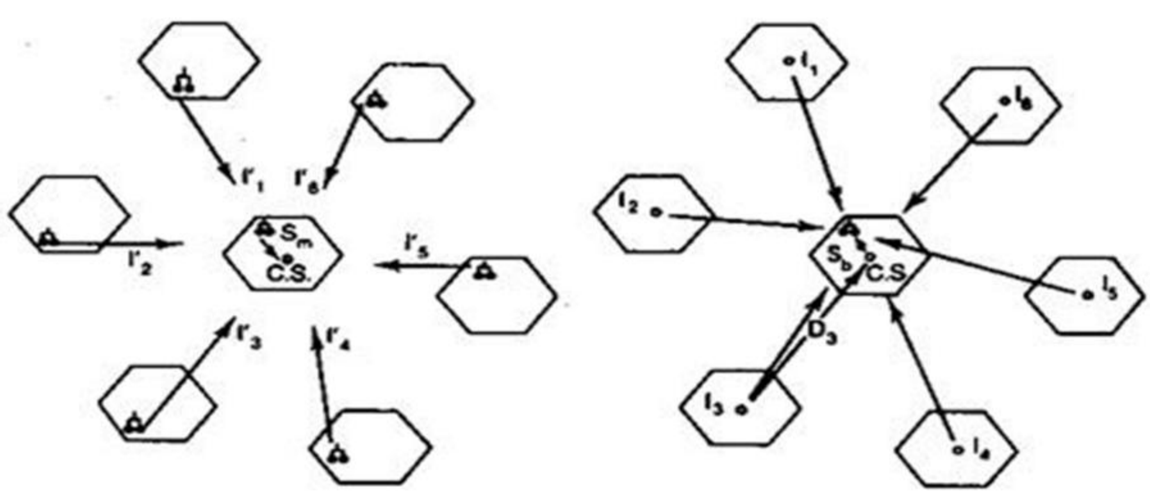


Fig 2.4 Co channel interference from six interferers. (a).receiving at the cell site; (b) receiving at the mobile unit.

Hand-off Mechanism: Hand-off is the process of automatically changing the frequencies. When the mobile unit moves out of the coverage areas of a particular cell site, the reception becomes weak. At this instant the present cell site requests Hand-off, then system switches the call to a new frequency channel in a new cell site without interrupting either call or user. This phenomenon is known as “hand -off’ or ‘handover’. Hand -off processing scheme is an important task for any successful mobile system. This concept can he applied to one dimensional as well as two dimensional cellular configurations.

By the reception of weak signals from the mobile unit by the cell site, the Hand-off is required in the following two situations. They are

The level for requesting a Hand-off in a noise limited environment is at the cell boundary say-l00 dBm.

In a particular cell site, when the mobile unit is reaching the signal strength holes (gaps).

Figure 2.5 shows the usage of frequency F1 in two co channel cells which arc separated by a distance D. Now, we have to provide a communication system in the whole area by filling other frequency channels F2,F3 and F4 between two co-channel cells. Depending on the same value of q the cells C2 ,C3 and C4 to which the above fill-in frequencies F2, F3 and F4 are assigned respectively as shown in figure.

Initially a mobile unit is starting a call in cell with fill-in frequency F1 and then moves to a cell with fill-in frequency F2. The mobile unit moves from cell C1 to cell C2, meanwhile however the call being dropped and reinitiated in the frequency channel from F1 to F2. This process of changing frequencies can be done automatically by the system without the user's intervention. In the cellular system the above mentioned Hand-off process is used.

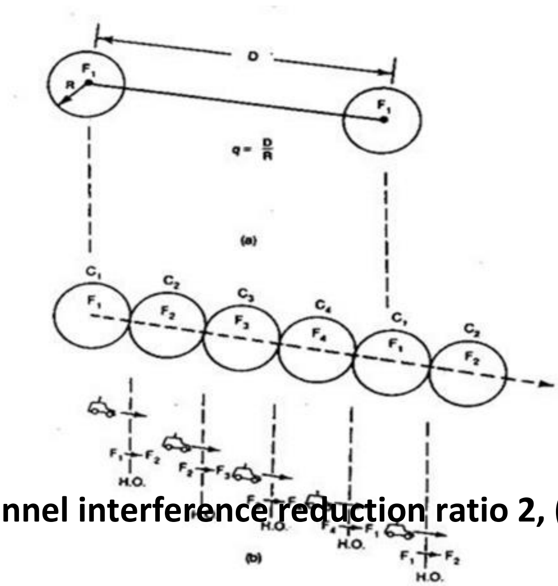


Fig.2.5: (a). co channel interference reduction ratio 2, (b) fill in frequency need of splitting

The motivation behind implementing a cellular mobile system is to improve the utilization of spectrum efficiency. The frequency reuse scheme is one concept, and cell splitting is another concept. When traffic density starts to build up and the frequency channels Fi in each cell Ci cannot provide enough mobile calls, the original cell can be split into smaller cells. Usually the new radius is one-half the original radius. There are two ways of splitting: In Fig. 8 a, the original cell site is not used, while in Fig. 8 b, it is

New cell radius = Old cell radius/2

Then,

New cell area = Old cell area/4

Let each new cell carry the same maximum traffic load of the old cell, then

New traffic load/Unit area = 4 X Traffic load/Unit area.

Splitting

There are two kinds of cell-splitting techniques:

Permanent splitting: The installation of every new split cell has to be planned ahead of time; the number of channels, the transmitted power, the assigned frequencies, the choosing of the cell-site selection, and the traffic load consideration should all be considered. When ready, the actual service cut-over should be set at the lowest traffic point, usually at midnight on a weekend. Hopefully, only a few calls will be dropped because of this cut-over, assuming that the downtime of the system is within 2 h.

Dynamic splitting: This scheme is based on using the allocated spectrum efficiency in real time. The algorithm for dynamically splitting cell sites is a tedious job, as we cannot afford to have one single cell unused during cell splitting at heavy traffic hours.

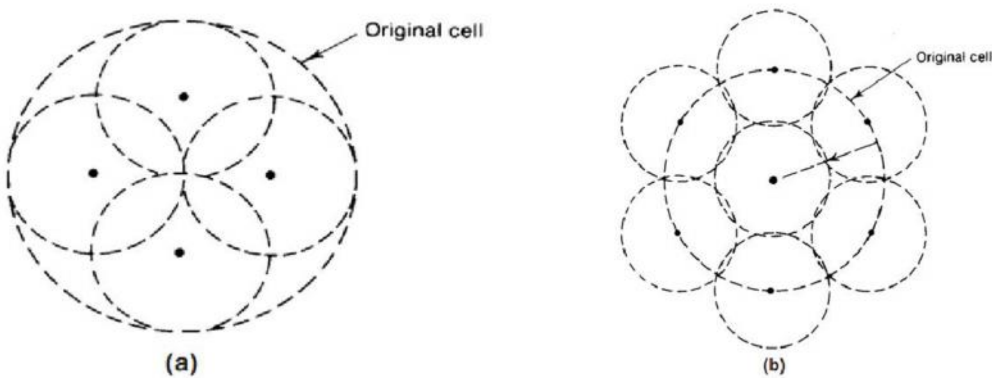


Fig.2.6: Cell splitting

the importance of $K = i^2 + ij + j^2$

For hexagonal cells i.e. with "honeycomb" cell layouts commonly used in mobile radio with possible cluster sizes are

$$K = i^2 + ij + j^2$$

Where i, j — Non negative integers

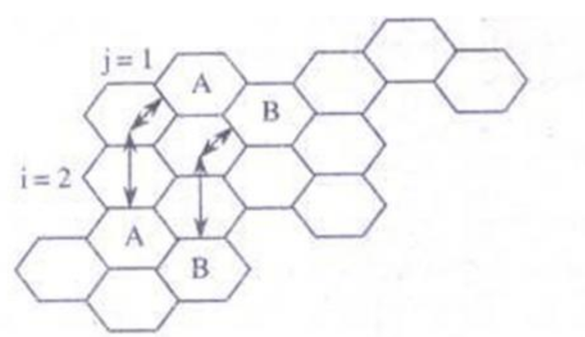
The integers i, j determine the relative location of co channels. The main reason for obtaining the above expression is to calculate the smallest number K which can still meet our system performance requirements. This process involves estimating co-channel interference and selecting the minimum frequency reuse distance D to reduce co channel interference. Thus, the smallest possible value for K is 3, obtained by putting i=1,j=1 in above eq.

The nearest co-channel neighbors of a particular cell can be obtained by the following two steps

(i) Moving i cells along any chain of hexagons.

(ii) Turn 60 degrees counter-clockwise and move j cells.

The method of locating co channel cells in a 7-cell reuse pattern with i=2 and j=1 is shown figure



The equation for frequency reuse pattern $K = i^2 + ij + j^2$ can also be used to measure the following.

I. The distance between co-channel cells in adjacent clusters is given by

$D = \sqrt{i^2 + ij + j^2}$.

The number of cells in a cluster, K is obtained by $N = D^2 = i^2 + ij + j^2$.

The frequency reuse factor, Q is obtained by

$$Q = \frac{D}{R} = \sqrt{3N} = \sqrt{3(i^2 + ij + j^2)}$$

HAND OFF

Why hand off is necessary ?

In an analog system, once a call is established, the set-up channel is not used again during the call period. Therefore, handoff is always implemented on the voice channel. In the digital systems, the handoff is carried out through paging or common control channel. The value of implementing handoffs is dependent on the size of the cell. For example, if the radius of the cell is 32 km (20 mi), the area is 3217 km^2(1256 mi^2). After a call is initiated in this area, there is little chance that it will be dropped before the call is terminated as a result of a weak signal at the coverage boundary. Then why bother to implement the handoff feature? Even for a 16-km radius, cell handoff may not be needed. If a call is dropped in a fringe area, the customer simply redials and reconnects the call. Today the size of cells becomes smaller in order to increase capacity. Also people talk longer. The handoffs are very essential. Handoff is needed in two situations where the cell site receives weak signals from the mobile unit:

(1) at the cell boundary, say, −100 dBm, which is the level for request

handoff in a noise-limited environment; and
(2) when the mobile unit is reaching the signal-strength holes (gaps) within the cell site as shown in Fig.1.

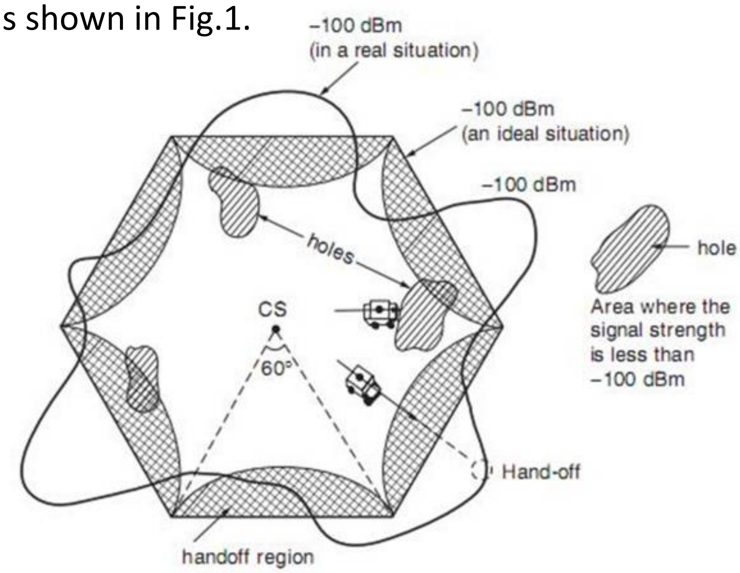


Fig7.1. Occurrence of handoffs

2. the two decision making parameters of handoff explain.

There are two decision-making parameters of handoff: (1) that based on signal strength and (2) that based on carrier-to-interference ratio. The handoff criteria are different for these two types. In type 1, the signal-strength threshold level for handoff is −100 dBm in noise-limited systems and −95 dBm in interference-limited systems. In type 2, the value of C/I at the cell boundary for handoff should be at a level, 18 dB for AMPS in order to have toll quality voice. Sometimes, a low value of C/I may be used for capacity reasons.

Type 1: It is easy to implement. The location receiver at each cell site measures all the signal strengths of all receivers at the cell site. However, the received signal strength (RSS) itself includes interference.

$$RSS = C + I$$

where C is the carrier signal power and I is the interference. Suppose that we set up a threshold level for RSS; then, because of the I , which is sometimes very strong, the RSS level is higher and far above the handoff threshold level. In this situation handoff should theoretically take place but does not. Another situation is when I is very low but RSS is also low. In this situation, the voice
quality usually is good even though the RSS level is low, but since RSS is low, unnecessary handoff takes place. Therefore, it is an easy but not very accurate method of determining handoffs. Some analog systems use SAT information together with the received signal level to determine handoffs. Some CDMA systems use pilot channel information.

Type 2: Handoffs can be controlled by using the carrier-to-interference ratio C/I

$$C+I/I = C/I$$

we can set a level based on C/I ,so C drops as a function of distance but I is dependent on the location. If the handoff is dependent on C/I , and if the C/I drops, it does so in response to increase in (1) propagation distance or (2) interference. In both cases, handoff should take place. In today's cellular systems, it is hard to measure C/I during a call because of analog modulation. Sometimes we measure the level I before the call is connected, and the level C + I during the call. Thus (C + I)/I can be obtained. Another method of measuring C/I is described in Sec. 9.3.

3. Concept of delaying a handoff

In many cases, a two-handoff-level algorithm is used. The purpose of creating two request handoff levels is to provide more opportunity for a successful handoff. A handoff could be delayed if no available cell could take the call. A plot of signal strength with two request handoff levels and a threshold level is shown in Fig.3. The plot of average signal strength is recorded on the channel received

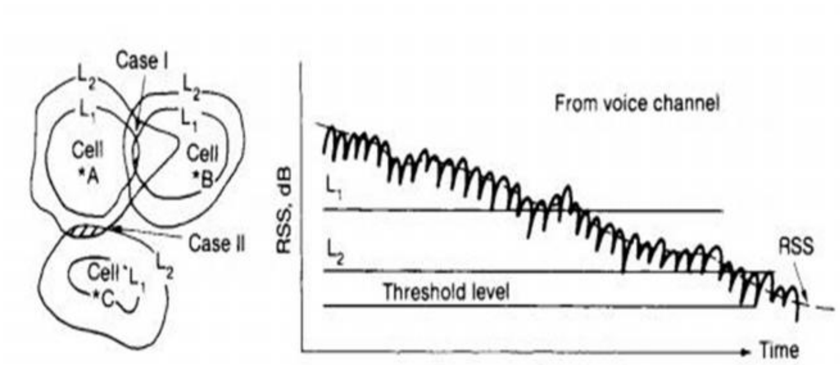


Fig.7.2. A two level handoff scheme

Signal strength indicator (RSSI), which is installed at each channel receiver at the cell site. When the signal strength drops below the first handoff level, a handoff request is initiated. If for some reason the mobile unit is in a hole (a weak spot in a cell) or a neighboring cell is busy, the handoff will be requested periodically every 5 s. At the first handoff level, the handoff takes place if the new signal is stronger. However, when the second handoff level is reached, the call will be handed off with no condition. The MSO always handles the handoff call first and the originating calls second. If no neighboring calls are available after the second handoff level is reached, the call continues until the signal strength drops below the threshold level; then the call is dropped. In AMPS systems if the supervisory audio tone (SAT) is not sent back to the cell site by the mobile unit within 5 s, the cell site turns off the transmitter.

4. The advantages of delayed handoff

Consider the following example. The mobile units are moving randomly and the terrain contour is uneven. The received signal strength at the mobile unit fluctuates up and down. If the mobile unit is in a hole for less than 5 s (a driven distance of 140 m for 5 s, assuming a vehicle speed of 100 km/h), the delay (in handoff) can even circumvent the need for a handoff. If the neighboring cells are busy, delayed handoff may take place. In principle, when call traffic is heavy, the switching processor is loaded, and thus a lower number of handoffs would help the processor handle call processing more adequately. Of course, it is very likely that after the second handoff level is reached, the call may be dropped with great probability. The other advantage of having a two-handoff-level algorithm is that it makes the handoff occur at the proper location and eliminates possible interference in the system.

Figure 7.2, case I, shows the area where the first-level handoff occurs between cell A and cell B. If we only use the second-level handoff boundary of cell A, the area of handoff is too close to cell B. Figure 3, case II, also shows where the second-level handoff occurs between cell A and cell C. This is because the first-level handoff cannot be implemented.

5. forced handoff

A forced handoff is defined as a handoff that would normally occur but is prevented from happening, or a handoff that should not occur but is forced to happen.

Controlling a Handoff:

The cell site can assign a low handoff threshold in a cell to keep a mobile unit in a cell longer or assign a high handoff threshold level to request a handoff earlier. The MSO also can control a handoff by making either a handoff earlier or later, after receiving a handoff request from a cell site.

Creating a Handoff:

In this case, the cell site does not request a handoff but the MSO finds that some cells are too congested while others are not. Then, the MSO can request call sites to create early handoffs for those congested cells. In other words, a cell site has to follow the MSO's order and increase the handoff threshold to push the mobile units at the new boundary and to handoff earlier.

Queuing of handoff:

Queuing of handoffs is more effective than two- threshold-level handoffs. The MSO will queue the requests of handoff calls instead of rejecting them if the new cell sites are busy. A queuing scheme becomes effective only when the requests for handoffs arrive at the MSO in batches or bundles. If handoff requests arrive at the MSO uniformly, then the queuing scheme is not needed. Before showing the equations, let us define the parameters as follows. $1/\mu$ average calling time in seconds, including new calls and handoff calls in each cell

$\lambda 1$	arrival rate ($\lambda 1$ calls per second) for originating calls
$\lambda 2$	arrival rate ($\lambda 2$ handoff calls per second) for handoff calls
M1	size of queue for originating calls
M2	size of queue for handoff calls
N	number of voice channels
a	$(\lambda 1 + \lambda 2)/\mu$
b1	$\lambda 1/\mu$
b2	$\lambda 2/\mu$

The following analysis can be used to see the improvement. We are analyzing three cases.

1. No queuing on either the originating calls or the handoff calls. The blocking for either an originating call or a handoff call is

$$B_o = \frac{a^N}{N!} P(0)$$

Where $$P(0) = \left(\sum_{n=0}^{N} \frac{a^N}{n!} \right)^{-1}$$

2. Queuing the originating calls but not the handoff calls. The blocking probability for originating calls is

$$B_{oq} = \left(\frac{b_1}{N} \right)^{M_1} P_q(0)$$

Where $$P_q(0) = \left[N! \sum_{n=0}^{N-1} \frac{a^{n-N}}{n!} + \frac{1 - (b_1/N)^{M_1+1}}{1 - (b_1/N)} \right]^{-1}$$

The blocking probability for handoff calls is

$$B_{oh} = \frac{1 - (b_1/N)^{M_1+1}}{1 - (b_1/N)} P_q(0)$$

3. Queuing the handoff calls but not the originating calls. The blocking probability for handoff calls is

$$B_{hq} = \left(\frac{b_2}{N} \right)^{M_2} P_q(0)$$

The blocking probability for originating calls is

$$B_{ho} = \frac{1 - (b_2/N)^{M_2+1}}{1 - (b_2/N)} P_q(0)$$

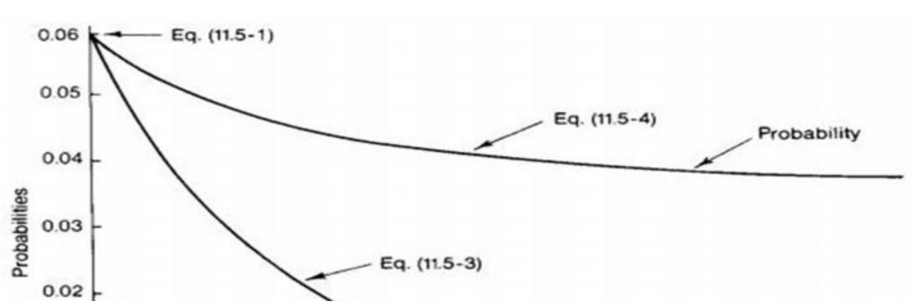


Fig.7.3Originating queue size

We have seen (Fig.7.3) with queuing of originating calls only, the probability of blocking is reduced. However, queuing of originating calls results in increased blocking probability on handoff calls, and this is a drawback. With queuing of handoff calls only, blocking probability is reduced from 5.9 to 0.1 percent by using one queue space. Therefore it is very worthwhile to implement a simple queue (one space) for handoff calls. Adding queues in handoff calls does not affect the blocking probability of originating calls.

However, we should always be aware that queuing for the handoff is more important than queuing for those initiating calls on assigned voice channels because call drops upset customers more than call blockings.

6. Power difference handoff

A better algorithm is based on the power difference ( ) of a mobile signal received by two cell sites, home and handoff. can be positive or negative. The handoff occurs depending on a preset value of .

= the mobile signal measured at the candidate handoff site

− the mobile signal measured at the home site

For example, the following cases can occur.

> 3 dB request a handoff

1dB < < 3 dB prepare a handoff

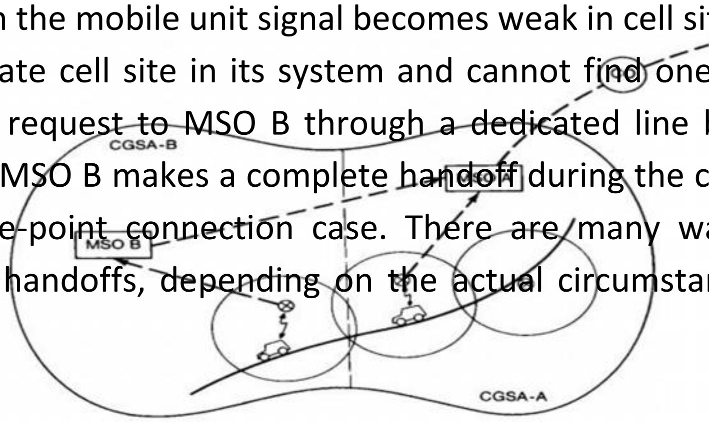
−3dB < < 0 dB monitoring the signal strength

< −3 dB no handoff

Those numbers can be changed to fit the switch processor capacity. This algorithm is not based on the received signal strength level, but on a relative (power difference) measurement. Therefore, when this algorithm is used, all the call handoffs for different vehicles can occur at the same general location in spite of different mobile antenna gains or heights.

7. Intersystem handoff

Occasionally, a call may be initiated in one cellular system (controlled by one MSO) and enter another system (controlled by another MSO) before terminating. In some instances, intersystem handoff can take place; this means that a call handoff can be transferred from one system to a second system so that the call is continued while the mobile unit enters the second system. The software in the MSO must be modified to apply this situation. Consider the simple diagram shown in Fig.7. The car travels on a highway and the driver originates a call in system A. Then the car leaves cell site A of system A and enters cell site B of system B. Cell sites A and B are controlled by two different MSOs. When the mobile unit signal becomes weak in cell site A, MSO A searches for a candidate cell site in its system and cannot find one. Then MSO A sends the handoff request to MSO B through a dedicated line between MSO A and MSO B, and MSO B makes a complete handoff during the call conversation. This is just a one-point connection case. There are many ways of implementing intersystem handoffs, depending on the actual circumstances. For instance, if



two MSOs are manufactured by different companies, then compatibility must be determined before implementation of intersystem handoff can be considered.

8. dropped call rate and consideration of dropped call rates

The definition of a dropped call is after the call is established but before it is properly terminated. The definition of "the call is established" means that the call is setup completely by the setup channel. If there is a possibility of a call drop due to no available voice channels, this is counted as a blocked call not a dropped call. If there is a possibility that a call will drop due to the poor signal of the assigned voice channel, this is considered a dropped call. This case can happen when the mobile or portable units are at a standstill and the radio carrier is changed from a strong setup channel to a weak voice channel due to the selective frequency fading phenomenon.

The perception of dropped call rate by the subscribers can be higher due to:

1The subscriber unit not functioning properly (needs repair).

2The user operating the portable unit in a vehicle (misused).

3. The user not knowing how to get the best reception from a portable unit (needs education).

In principle, dropped call rate can be set very low if we do not need to maintain the voice quality. The dropped call rate and the specified voice quality level are inversely proportional. In designing a commercial system, the specified voice quality level is given relating to how much C/I (or C/N) the speech coder can tolerate. By maintaining a certain voice quality level, the dropped call rate can be calculated by taking the following factors into consideration:

1. Provide signal coverage based on the percentage (say 90 percent) that the entire

received signal will be above a given signal level.

2. Maintain the specified co-channel and adjacent channel interference levels in each cell during a busy hour (i.e., the worst interference case).

3. Because the performance of the call dropped rate is calculated as possible call dropping in every stage from the radio link to the PSTN connection, the response time of the handoff in the network will be a factor when the cell becomes small, the response time for a handoff request has to be shorter in order to reduce the call dropped rate.

9. Relation among capacity, voice quality, dropped call rate

Radio Capacity m is expressed as follows:

$$m = \frac{B_T / B_c}{\sqrt{\frac{2}{3}(C/I)_S}}$$

Where BT/Bc is the total number of voice channels. BT /Bc is a given number, and (C/I)s is a required C/I for designing a system. The above equation is obtained based on six co channel interferers which occur in busy traffic (i.e., a worst case). In an interference limited system, the adjacent channel interference has only a secondary effect.

$$(C/I)_S = \frac{3}{2}\left(\frac{B_T/B_c}{m}\right)^2 = \frac{3}{2}\left(\frac{B_T}{B_c}\right)^2 \cdot \frac{1}{m^2}$$

Because the (C/I)s is a required C/I for designing a system, the voice quality is based on the (C/I)s .When the specified (C/I)S is reduced, the radio capacity is increased. When the measured (C/I) is less than the specified (C/I)S , both poor voice quality and dropped calls can occur.

General formula of dropped call rate

The general formula of dropped call rate P in a whole system can be expressed as:

$$P = 1 - \left[\sum_{n=0}^{N} \alpha_n X^n\right] = \sum_{n=0}^{N} \alpha_n \cdot P_n$$

Where $$P_n = 1 - X^n$$

P_n is the probability of a dropped call when the call has gone through n handoffs and

$X = (1-\delta)(1-\mu)(1-\theta\tau)(1-\beta)^2$

δ = Probability that the signal is below the specified receive threshold (in a noise-limited system).

μ = Probability that the signal is below the specified cochannel interference level (in an interference-limited system).

τ = Probability that no traffic channel is available upon handoff attempt when moving into a new cell.

θ = Probability that the call will return to the original cell.

β = Probability of blocking circuits between BSC and MSC during handoff.

α_n = The weighted value for those calls having n handoffs, and $\sum_{n=0}^{N} \alpha_n = 1$

N = N is the highest number of handoffs for those calls.

1. z1 and z2 are two events, z1 is the case of no traffic channel in the cell, z2 is the case of no-safe return to original cell. Assuming that z1 and z2 are independent events, then
2. (1 − β) is the probability of a call successfully connecting from the old BSC to the MSC. Also, (1 − β) is the probability of a call successfully connecting from the MSC to the new BSC. Then the total probability of having a successful call connection is

   The call dropped rate P expressed in above Eq can be specified in two cases:

1. In a noise limited system (startup system): there is no frequency reuse, the call dropped rate PA is based on the signal coverage. It can also be calculated under busy hour conditions.

In a noise-limited environment (for worst case)

$$\delta = \delta_1$$

$$\mu = \mu_1$$

$$\left.\begin{aligned} \tau &= \tau_1 \\ \theta &= \theta_1 \\ \beta &= \beta_1 \end{aligned}\right\} \text{the conditions for the noise limited case}$$

2. In an interference-limited system (mature system): frequency reuse is applied, and the dropped rate PB is based on the interference level. It can be calculated under busy hour conditions.

In an interference-limited environment (for worst case)

$$\delta = \delta_2$$

$$\mu = \mu_1$$

$$\left.\begin{aligned} \tau &= \tau_2 \\ \theta &= \theta_2 \\ \beta &= \beta_2 \end{aligned}\right\} \text{the conditions for the interference limited case}$$

In a commonly used formula of dropped call rate, the values of τ , θ, and β are assumed to be very small and can be neglected. Then

$$X = (1 - \delta)(1 - \mu)$$

Furthermore, in a noise-limited case, $\mu \to 0$,

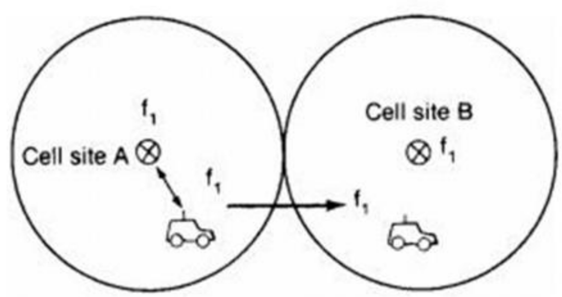
$$P_A = \sum_{n=0}^{N} \alpha_n P_n = \sum \alpha_n [1 - (1 - \delta)^n]$$

and in an interference-limited system, $\delta \to 0$,

$$P_B = \sum_{n=0}^{N} \alpha_n P_n = \sum \alpha_n [1 - (1 - \mu)^n]$$

Cell site handoff scheme:

This scheme can be used in a non cellular system. The mobile unit has been assigned a frequency and talks to its home cell site while it travels. When the mobile unit leaves its home cell and enters a new cell, its frequency does not change; rather, the new cell must tune into the frequency of the mobile unit . In this case only the cell sites need the frequency information of the mobile unit. Then the aspects of mobile unit control can be greatly simplified, and there will be no need to provide handoff capability at the mobile unit. The cost will also be lower. This scheme can be recommended only in areas of very low traffic. When the traffic is dense, frequency coordination is necessary for the cellular system. Then if a mobile unit does not change frequency on travel from cell to cell, other mobile units then must change frequency to avoid interference. Therefore, if a system handles only low volumes of traffic, that is, if the channels assigned to one cell will not reuse frequency in other cells, then it is possible to implement the cell-site handoff feature as it is applied in military systems.



Cell site handoff only scheme

$$P(z_2|z_1) \cdot P(z_1) = P(z_2) \cdot P(z_1) = \theta \cdot \tau$$

$$\left.\begin{array}{ll} \text{BSC (old)} \rightarrow \text{MSC} & (1-\beta) \\ \text{MSC} \rightarrow \text{BSC (new)} & (1-\beta) \end{array}\right\} \rightarrow (1-\beta)^2$$

Multiple Access Techniques

Multiple access techniques are used to allow a large number of mobile users to share the allocated spectrum in the most efficient manner. As the spectrum is limited, so the sharing is required to increase the capacity of cell or over a geographical area by allowing the available bandwidth to be used at the same time by different users. And this must be done in a way such that the quality of service doesn't degrade within the existing users.

Multiple Access Techniques for Wireless Communication

In wireless communication systems it is often desirable to allow the subscriber to send simultaneously information to the base station while receiving information from the base station.

A cellular system divides any given area into cells where a mobile unit in each cell communicates with a base station. The main aim in the cellular system design is to be able to increase the capacity of the channel i.e. to handle as many calls as possible in a given bandwidth with a sufficient level of quality of service. There are several different ways to allow access to the channel. These includes mainly the following:

1) Frequency division multiple-access (FDMA)
2) Time division multiple-access (TDMA)
3) Code division multiple-access (CDMA)
4) Space Division Multiple access (SDMA)

Table: MA techniques in different wireless communication systems

Advanced Mobile Phone Systems:	FDMA/FDD
Global System for Mobile:	TDMA/FDD
U.S. Digital Cellular:	TDMA/FDD
Japanese Digital Cellular:	TDMA/FDD

CT2 Cordless Telephone:	FDMA/TDD
Digital European Cordless Telephone:	FDMA/TDD
U.S. Narrowband Spread Spectrum(IS-95):	CDMA/FDD

FDMA,TDMA and CDMA are the three major multiple access techniques that are used to share the available bandwidth in a wireless communication system. Depending on how the available bandwidth is allocated to the users these techniques can be classified as narrowband and wideband systems.

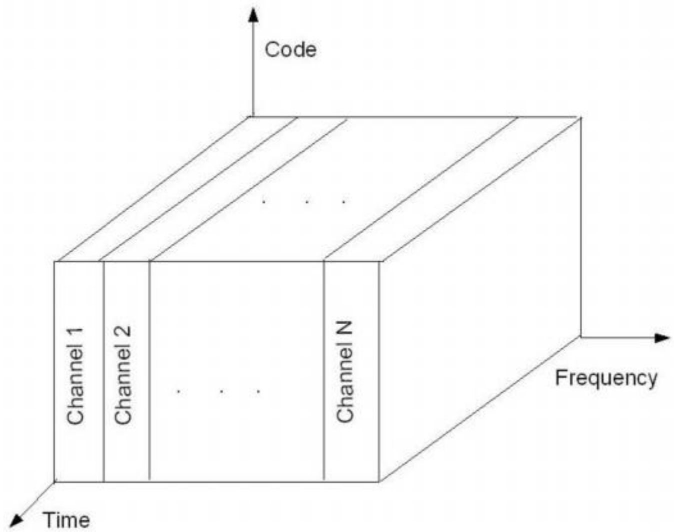
Narrowband Systems

The term narrowband is used to relate the bandwidth of the single channel to the expected coherence bandwidth of the channel. The available spectrum is divided in to a large number of narrowband channels. The channels are operated using FDD. In narrow band FDMA, a user is assigned a particular channel which is not shared by other users in the vicinity and if FDD is used then the system is called FDMA/FDD. Narrow band TDMA allows users to use the same channel but allocated a unique time slot to each user on the channel, thus separating a small number of users in time on a single channel. For narrow band TDMA, there generally are a large number of channels allocated using either FDD or TDD, each channel is shared using TDMA. Such systems are called TDMA/FDD and TDMA/TDD access systems.

Wideband Systems

In wideband systems, the transmission bandwidth of a single channel is much larger than the coherence bandwidth of the channel. Thus, multipath fading doesn't greatly affect the received signal within a wideband channel, and frequency selective fades occur only in a small fraction of the signal bandwidth

Figure 2.1: The basic concept of FDMA.



2.2 Frequency Division Multiple Access

This was the initial multiple-access technique for cellular systems in which each individual user is assigned a pair of frequencies while making or receiving a call as shown in Figure1. One frequency is used for downlink and one pair for uplink. This is called frequency division duplexing (FDD). That allocated frequency pair is not used in the same cell or adjacent cells during the call so as to reduce the co channel interference. Even though the user may not be talking, the spectrum cannot be reassigned as long as a call is in place. Different users can use the same frequency in the same cell except that they must transmit at different times.

The features of FDMA are as follows: The FDMA channel carries only one phone circuit at a time. If an FDMA channel is not in use, then it sits idle and it cannot be used by other users to increase share capacity. After the assignment of the voice channel the BS and the MS transmit simultaneously and continuously.

The bandwidths of FDMA systems are generally narrow i.e. FDMA is usually implemented in a narrow band system The symbol time is large compared to the average delay spread. The complexity of the FDMA mobile systems is lower than that of TDMA mobile systems. FDMA requires tight filtering to minimize the adjacent channel interference.

2.2.1 FDMA/FDD in AMPS

The first U.S. analog cellular system, AMPS (Advanced Mobile Phone System) is based on FDMA/FDD. A single user occupies a single channel while the call is in progress, and the single channel is actually two simplex channels which are frequency duplexed with a 45 MHz split. When a call is completed or when a handoff occurs the channel is vacated so that another mobile subscriber may use it. Multiple or simultaneous users are accommodated in AMPS by giving each user a unique signal.

Voice signals are sent on the forward channel from the base station to the mobile unit, and on the reverse channel from the mobile unit to the base station. In AMPS, analog narrowband frequency modulation (NBFM) is used to modulate the carrier.

2.2.2 FDMA/TDD in CT2

Using FDMA, CT2 system splits the available bandwidth into radio channels in the assigned frequency domain. In the initial call setup, the handset scans the available channels and locks on to an unoccupied channel for the duration of the call. Using TDD(Time Division Duplexing), the call is split into time blocks that alternate between transmitting and receiving.

2.2.3 FDMA and Near-Far Problem

The near-far problem is one of detecting or filtering out a weaker signal amongst stronger signals. The near-far problem is particularly difficult in CDMA systems where transmitters share transmission frequencies and transmission time. In contrast, FDMA and TDMA systems are less vulnerable. FDMA systems offer different kinds of solutions to near-far challenge. Here, the worst case to consider is recovery of a weak signal in a frequency slot next to strong signal. Since both signals are present simultaneously as a composite at the input of a gain stage, the gain is set according to the level of the stronger signal; the weak signal could be lost in the noise floor. Even if subsequent stages have a low enough noise floor to provide

2.3 Time Division Multiple Access

In digital systems, continuous transmission is not required because users do not use the allotted bandwidth all the time. In such cases, TDMA is a complimentary access technique to FDMA. Global Systems for Mobile communications (GSM) uses the TDMA technique. In TDMA, the entire bandwidth is available to the user but only for a finite period of time. In most cases the available bandwidth is divided into fewer channels compared to FDMA and the users are allotted time slots during which they have the entire channel bandwidth at their disposal, as shown in Figure(2.2)

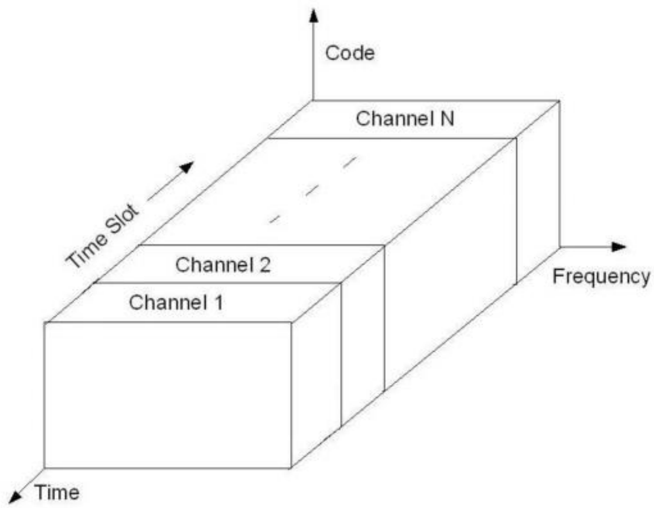
TDMA requires careful time synchronization since users share the bandwidth in the frequency domain. The number of channels are less, inter channel interference is almost negligible. TDMA uses different time slots for transmission and reception. This type of duplexing is referred to as Time division duplexing(TDD).

The features of TDMA includes the following: TDMA shares a single carrier frequency with several users where each users makes use of non overlapping time slots.The number of time slots per frame depends on several factors such as modulation technique, available bandwidth etc. Data transmission in TDMA is not continuous but occurs in bursts. This results in low battery consumption since the subscriber transmitter can be turned OFF when not in use. Because of a discontinuous transmission in TDMA the handoff process is much simpler for a subscriber unit, since it is able to listen to other base stations during idle time slots. TDMA uses different time slots for transmission and reception thus duplexers are not required. TDMA has an advantage that is possible to allocate different numbers of time slots per frame to different users. Thus bandwidth can be supplied on demand to different users by concatenating or reassigning time slot based on priority.

2.3.1 TDMA/FDD in GSM

As discussed earlier, GSM is widely used in Europe and other parts of the world. GSM uses a variation of TDMA along with FDD. GSM digitizes and compresses data, then sends it down a channel with two other streams of user data, each in its

Figure 2.2: The basic concept of TDMA.



own time slot. It operates at either the 900 MHz or 1800 MHz frequency band. Since many GSM network operators have roaming agreements with foreign operators, users can often continue to use their mobile phones when they travel to other countries.

8.3.2 TDMA/TDD in DECT

DECT is a pan European standard for the digitally enhanced cordless telephony using TDMA/TDD. DECT provides 10 FDM channels in the band 1880-1990 Mhz. Each channel supports 12 users through TDMA for a total system load of 120 users. DECT supports handover, users can roam over from cell to cell as long as they remain within the range of the system. DECT antenna can be equipped with optional spatial diversity to deal with multipath fading.

2.4 Spread Spectrum Multiple Access

Spread spectrum multiple access (SSMA) uses signals which have a transmission bandwidth whose magnitude is greater than the minimum required RF bandwidth. A pseudo noise (PN) sequence converts a narrowband signal to a wideband noise like signal before transmission. SSMA is not very bandwidth efficient when used by a single user. However since many users can share the same spread spectrum bandwidth without interfering with one another, spread spectrum systems become bandwidth efficient in a multiple user environment.

There are two main types of spread spectrum multiple access techniques:
Frequency hopped multiple access (FHMA)
Direct sequence multiple access (DSMA) or
Code division multiple access (CDMA).

2.4.1 Frequency Hopped Multiple Access (FHMA)

This is a digital multiple access system in which the carrier frequencies of the individual users are varied in a pseudo random fashion within a wideband channel. The digital data is broken into uniform sized bursts which is then transmitted on different carrier frequencies.

2.4.2 Code Division Multiple Access

In CDMA, the same bandwidth is occupied by all the users, however they are all assigned separate codes, which differentiates them from each other (shown in Figure2.3). CDMA utilize a spread spectrum technique in which a spreading signal (which is uncorrelated to the signal and has a large bandwidth) is used to spread the narrow band message signal.

Direct Sequence Spread Spectrum (DS-SS)

This is the most commonly used technology for CDMA. In DS-SS, the message signal is multiplied by a Pseudo Random Noise Code. Each user is given his own codeword which is orthogonal to the codes of other users and in order to detect the user, the receiver must know the codeword used by the transmitter. There are, however, two problems in such systems which are discussed in the sequel.

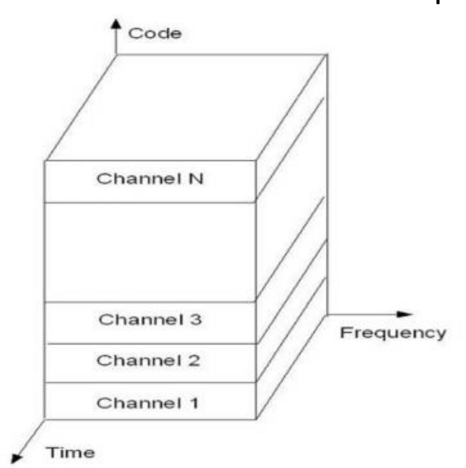


Figure 2.3: The basic concept of CDMA.

CDMA/FDD in IS-95

In this standard, the frequency range is: 869-894 MHz (for Rx) and 824-849 MHz (for Tx). In such a system, there are a total of 20 channels and 798 users per channel. For each channel, the bit rate is 1.2288 Mbps. For orthogonality, it usually combines 64 Walsh-Hadamard codes and a m-sequence.

2.4.3 CDMA and Self-interference Problem

In CDMA, self-interference arises from the presence of delayed replicas of signal due to multipath. The delays cause the spreading sequences of the different users to lose their orthogonality, as by design they are orthogonal only at zero phase offset. Hence in despreading a given user's waveform, nonzero contributions to that user's signal arise from the transmissions of the other users in the network. This is distinct from both TDMA and FDMA, wherein for reasonable time or frequency guard bands,respectively, orthogonality of the received signals can be preserved.

2.4.4 CDMA and Near-Far Problem

The near-far problem is a serious one in CDMA. This problem arises from the fact that signals closer to the receiver of interest are received with smaller attenuation than are signals located further away. Therefore the strong signal from the nearby transmitter will mask the weak signal from the remote transmitter. In TDMA and FDMA, this is not a problem since mutual interference can be filtered. In CDMA, however, the near-far effect combined with imperfect orthogonality between codes (e.g. due to different time sifts), leads to substantial interference. Accurate and fast power control appears essential to ensure reliable operation of multiuser DS-CDMA systems.