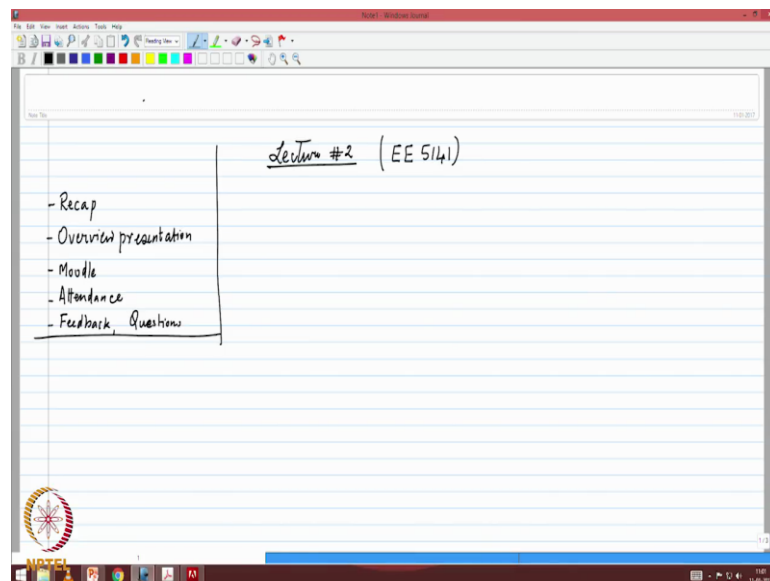


Introduction to Wireless and Cellular Communication
Prof. David Koilpillai
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Lecture - 02
Overview of Cellular Evaluation and Wireless Technologies
Overview of Cellular Systems – Part 2

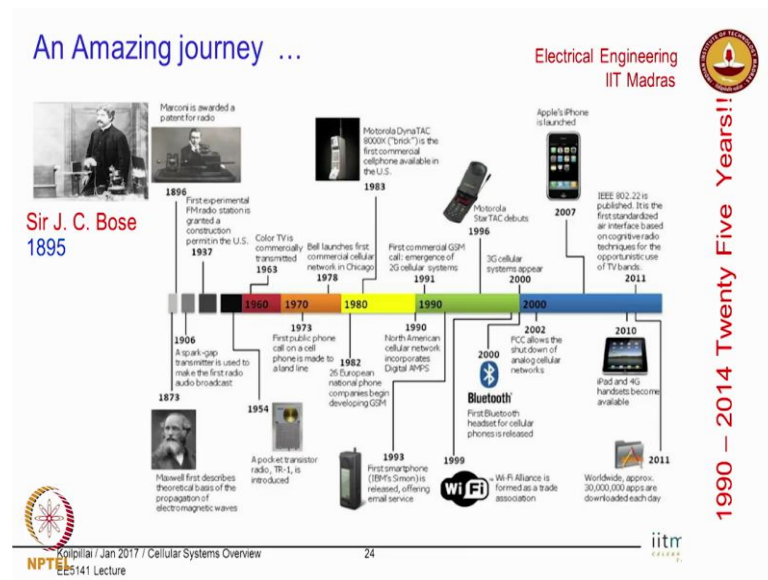
Welcome. Lecture number 2 we will continue the overview presentation. I hope you had a chance to look at the slides the full presentation was uploaded in module yesterday at least had a chance to look at the slides that we covered yes in the class. There is a certain pattern how we are trying to cover a little bit new little bit old looking at mixing it up, making sure that we connect to what you have already studied in a digital communications.

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So, we will continue the presentation, but interrupting it here and there for some discussion points. So, let me start with a slide that I believe is a good recap of what we have discussed yesterday.

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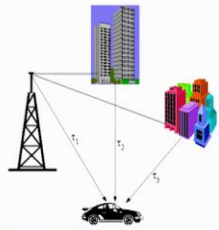


So, this is a pictorial representation of the evolution of the wireless systems Maxwell of course, the father of electromagnetic theory, the foundations for wireless communication. But J. C Bose proceeding Marconi's work the first digital cellular systems 1991 you know probably remember it from the presentation yesterday then came along Wi-Fi, then Bluetooth, then 3G and then the rest of it. So, much of the exciting work in cellular and wireless has happened in the last 25 years, so interesting aspect that we can keep in mind.

The key points that we had mentioned yesterday in the discussion was that our course is going to focus on the, our understanding of the wireless making sure that we have a good analytical understanding as well as a physical understanding as an intuition, so all 3 elements.

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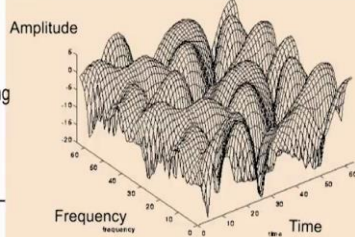
Multipath & Delay-spread




- Multipath \Rightarrow multiple copies of signal at receiver
- Fading \Rightarrow signal strength fluctuation
- Spatio-Temporal pattern (random)
- Receiver must adapt to signal fluctuations
- Loss of signal during deep fade

- Delay-spread
 - Paths arriving at different times
 - Inter-symbol interference (ISI)
- Need advanced DSP algorithms to mitigate time-varying ISI
 - Equalizer, RAKE receiver
 - Computationally intensive

Equalizer complexity grows exponentially



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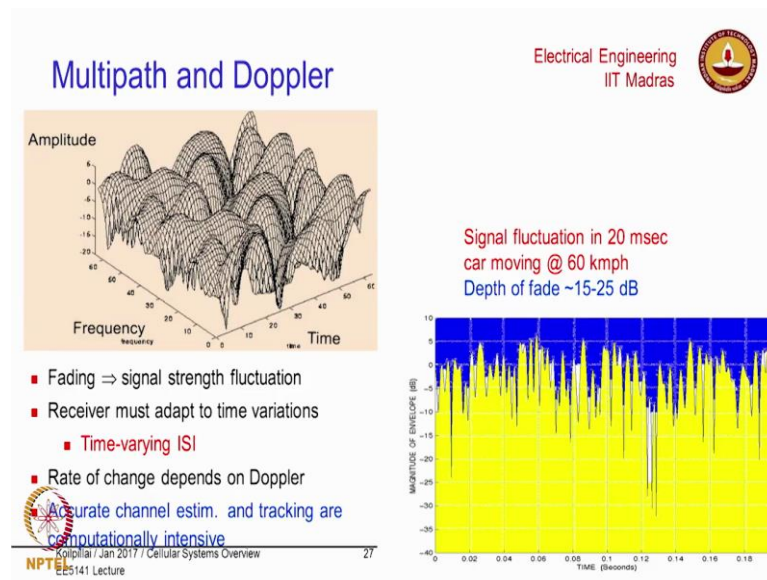


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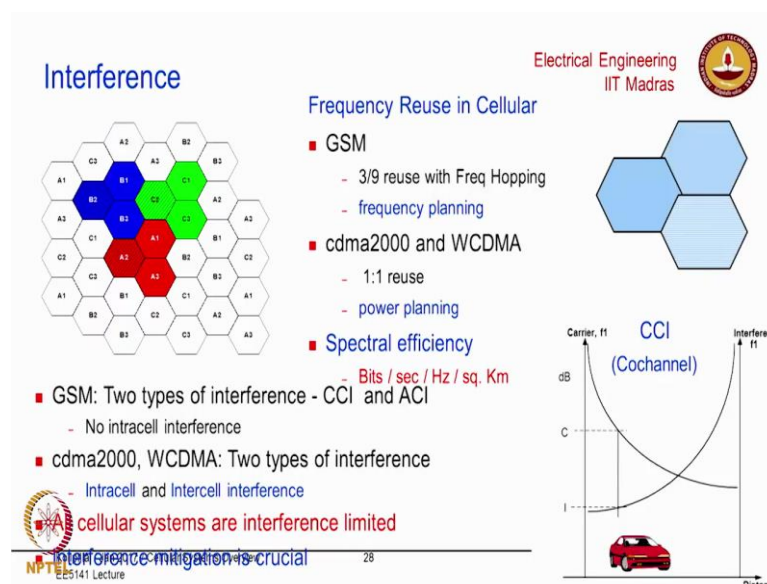
So, the points to remember is multipath propagation, the fact that these paths will cause a fluctuation of the signal from a represent, physical interpretation it is a spatiotemporal process, but from an analytical tool it just says that you know we think of communications as something that happens as a function of time this is a channel that is varying as a function of time, but actually it is a spatio temporal phenomenon what we observe is the impact on the time scale.

Then we also talked about the fact that these multipath components could come delayed in time that causes inter symbol interference. And the fact that if you are moving then the entire thing is changing, the received signal is changing, the multipath components are changing. So, this becomes part of the challenge that we have.

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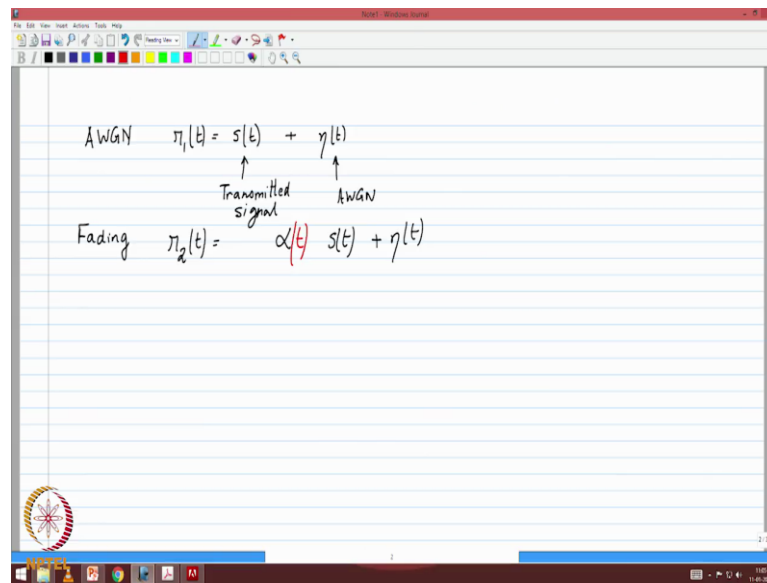


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Now, to add to that we also have interference two types of interference, inter cell interference - somebody from another cell who is using the same a frequency resources this person is not orthogonal in fact, they are perfectly correlated in terms of the resource. The other one is those who are within your own cell, who are supposed to be orthogonal, but who have for what one reason or the other have lost orthogonality and therefore, are causing interference. So, inter cell, intra cell, interference is also a phenomenon that we have to deal with. So, let me give you something to think about as we look at this.

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The image shows a digital notepad with two equations written on it. The first equation is for an Additive White Gaussian Noise (AWGN) channel: $r_1(t) = s(t) + \eta(t)$. Arrows point from the labels 'Transmitted signal' and 'AWGN' to $s(t)$ and $\eta(t)$ respectively. The second equation is for a Fading channel: $r_2(t) = \alpha(t) s(t) + \eta(t)$. An arrow points from the label 'Fading' to the equation. The term $\alpha(t)$ is written in red ink. The notepad has a standard toolbar at the top and a taskbar at the bottom.

$$\text{AWGN} \quad r_1(t) = s(t) + \eta(t)$$

↑ ↑
Transmitted signal AWGN

$$\text{Fading} \quad r_2(t) = \alpha(t) s(t) + \eta(t)$$

So, it is this what we represent as an AWGNs channel what I transmit is s of t , what I receive is s of t plus a noise and the level of the noise determines what is my signal to noise ratio, what is the probability that I will get a symbol error.

Now, the introduction of fading, let us call that as a channel which is r_2 of t , says that the AWGN is still present I have not done anything with that transmitted signal s of t now gets scaled by a factor α . And this α can be sometimes greater than 1 which is an which is a good thing you get better signal noise ratio, but most of the time you are worried about the scenarios when α is less than 1 and some cases significantly less than 1 causing you to have very poor signal to noise ratio.

Now given that we are talking about a spatio temporal process where the receiver can be moving then we say that this is actually a function of time as you move around. So, basically it becomes a time varying channel where your signal strength is constantly fluctuating and therefore, causing disturbance to the communication. Now how do we capture the fact that you can have multipath, the fact that you can have the desired signal plus some versions of the delayed versions of the signal.

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The image shows a digital whiteboard with handwritten notes. The notes are organized into three sections: 'Fading + ISI', 'Fading + interference', and 'Task'.

Fading + ISI

$$r_3(t) = \alpha_0(t) s(t) + \underbrace{\sum_{i=1}^L \alpha_i(t) s(t - \tau_i)}_{\text{ISI}} + \eta(t)$$

Fading + interference

$$r_4(t) = \alpha_0(t) s(t) + \underbrace{\beta_1(t) i_1(t)}_{\text{interference}} + \eta(t)$$

Task

1. K interferers
2. Desired + K interferers
(L -taps) L_1, L_2, \dots, L_K paths of $s(t)$

So, fading and ISI if we were to capture it that would become $\alpha_0(t)$ which is your basic signal, then $\eta(t)$ which is your AWGN and then the different ISI terms, this is the inter symbol interference caused by multipath components that are coming delayed in time.

Now, that is one type of interference, let us say that there is no inter symbol interference I only have the impact of the other sources of interference, co channel interference or the fact that I could have our intra cell interference. So, how would we capture that? So, basically this is a representation which says that my received signal is desired signal which is fluctuating because of the fading phenomenon that is happening, plus I have an interfering signal $i(t)$, now $i(t)$ also is a receiver as a signal that whose origin is not fixed. So, therefore, you can visualize it as a waiting function which is $\beta_1(t)$ let me call it as $i_1(t)$, this is 1 interfere plus the AWGN. So, this is my interference term. And whether it comes from co channel inter cell in does not matter this is some form of interference which is a disturbing my signal.

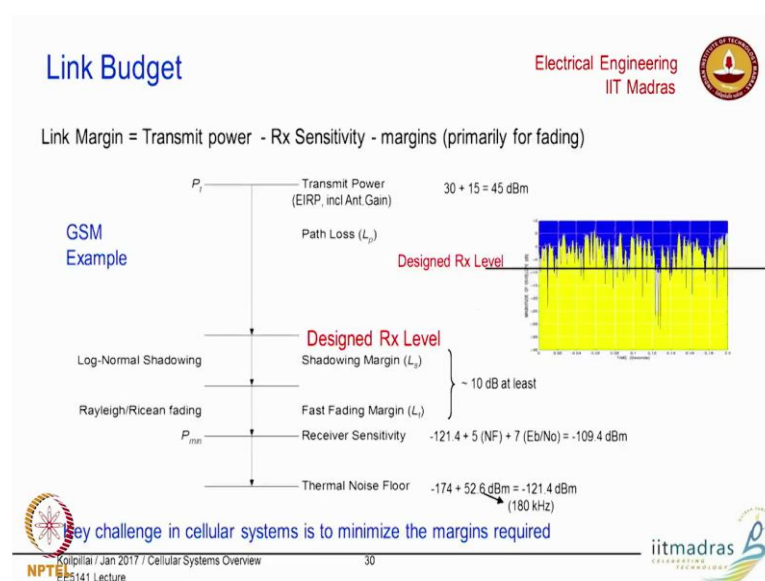
So, I hope you see that the characterization of the inter symbol interference, the characterization of the interference which is coming from other users not from your own signal, but from other signals can also be captured in this form. So, what I would like you to do is a task, the task is let us take the case where we have interference generate interference from K interferers, that is very straightforward it will now be a summation

in the interference term you will have β_1 to β_K and I want to β_K that will give you the interference.

Now, the second element which hopefully is something that you may have to think about is think of desired signal plus K interferers desired signal plus K interferers now each of them are passing through a dispersal channel. So, for the desired signal I have L taps of inter symbol interference, and for each of these K interference there are L_1, L_2 up to L_K paths of ISI. Please capture all of this into a single equation; that means, received signal is desired signal plus L taps of ISI, K interferers each of them are time dispersed. So, there are different multiple path components and you have AWGN.

So, in a nutshell that is the channel that we have to work with. So, as we go along we will ask the questions how do I get rid of ISI, how do I get rid of interference, and combine all of this together is a ability to design and build a very robust wireless system. Any questions, any questions? Feel free to ask because again I just have to make sure that whatever I have written is easily legible for you, if there is any doubts. So, now, I would like to go back to our earlier discussion and pick it up from there. The design of a wireless communication system which we are familiar with I just want to build on our understanding of the aspects of link budget.

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So, the link budget in essence tells you how much transmit power you should use to have a certain signal to noise ratio at the receiver. I want to sort of quickly run through the

details will be covered in a subsequent lecture discussion this is more of a conceptual explanation. So, thermal noise floor this is what is based on the electronic noise this has you have something you do not have a any control over what is the thermal noise dependent on; dependent on the Boltzmann's constant, the ambient temperature and the bandwidth of the receive filter that you have used. So, it is $k t b$.

So, maybe if you can write it down the thermal noise floor is based on $k t b$ and the numbers that you see in the chart I hope you will be able to verify we will give you the reference values for the Boltzmann's constant ambient temperature is taken as 300 Kelvin, and you can do the calculation and verify that you will come out with a number using a bandwidth of 180 kilohertz you should get 121.4, minus 121.4 dBm. There is a conversion from dBw to dBm I am assuming that a all communications engineers know how to do that that it is just a factor of 30 that has to be added to make it into dBm.

Now, that is really not the as point that we are interested in, we are interested in is what is the signal level that you must receive in order to be able to have error free or as close to error free transmission as possible that is called receiver sensitivity. The difference between the thermal noise floor which is a constant that you do not have any control over to what is the receiver sensitivity is controlled by two parameters, one is the noise figure of the receiver that you are using the poorer the noise figure; that means, more noise is being added by your receivers electronics, so therefore, you have to have higher received signal power. So, the noise figure usually a term that is it represented in dB is in this case of assumed to be 5 dB it is a fairly good receiver, poor receiver would have noise figure of may be 9 or 10, noise figure of 5 is considered good, noise figure of 2 would be even better; most commercial receivers would be in the 5 to 8 range noise figure in dB.

So, the noise figure, that the noise floor time plus the noise figure remember we are doing it all in decibels. So, therefore, it is a multiplicative factor in the expression, but in the decibel scale it becomes an additive factor and then that is your noise level, now from there you must have the minimum signal to noise ratio that you need. So, if your modulation scheme says you need 7 dB signal to noise ratio, it is whatever is the noise level plus 70 dB. So, noise floor plus noise figure plus E_b by naught required E_b by naught gives you the receiver sensitivity, and that is a very very important number in digital communications we may not even pay attention to it because we are interested only in a b e r and E_b by n naught, but when you build a system the first parameter that a

engineer will ask is what is the receiver sensitivity because you can only design your cell size such that you not making sure that the majority of the receivers can receive the transmitted signal from the base station at or better than minus 1 or 9.4 this particular example.

Now, in digital communications the way we would have done it is the up to this point is good then the only other thing that we will have to account for is path loss because you transmit with a certain power, that is propagation through the wireless medium then that comes down to. So, that would be a somewhat simpler link budget, now what is different in the context of a wireless system. If I designed it like that basically the receiver sensitivity plus path loss that is my transmit power, what would happen? The problem that I would run into is if there was no fading everything would be fine, the minute I have fading what will happen the signal will drop below the threshold I will start getting errors.

So, what do wireless systems designers do they say you have to have margin a buffer. So, we say that I need to have some buffer for fast fading basically as you are moving signal will fluctuate you need to have a buffer for that. Now in addition to that the user may go inside a building into a basement into an elevator so many things where the signal level itself will drop, that is called shadowing; that means, there is it is not because of fading it is because you have obstructions between you and the transmitter. So, that is called shadowing. So, basically there are two margins that are used one is fast fading margin usually so, L_f shadowing is L_s typically if you could add 20 dB to your link budget that would be ideal very very good because one concrete floor is actually 10 dB of margin. But again the price at you will have to pay if you make this 20 dB your transmit power will increase by 20 dB, your cost of your base station is going to increase. So, all operators want to keep this margin at the optimum level.

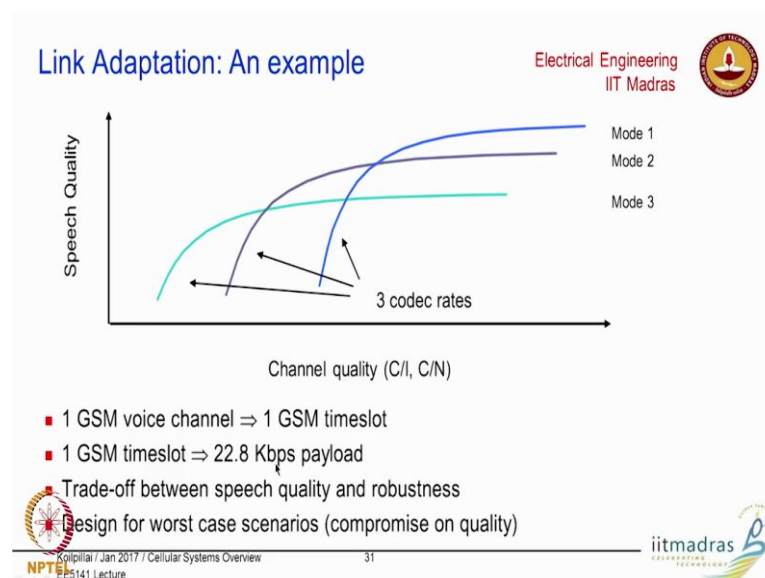
What is optimum? As small as possible so that the transmit power is manageable as big as possible so that you do not have problems due to fading or due to shadowing. So, the key element in a cellular system is how much margin you have given to your subscriber; the greater the margin the better more satisfied your subscribers will be because they will not see the effects of fading, and they will be very happy with the coverage. The lesser the margin that the more they are going to complain there will be dropped calls there will you know the call quality will not be good, so though. So, this is a very very important

slide understanding the fact that margins are a very key part of our cellular system design and a ability to a manipulate that is or is to work with that is it is very very good.

Now, here is an observation, let us say I have put in 20 dB of margin and I am in good signal conditions my signal to noise ratio is 20 dB more than what it should be because that is the positive effect of the margin. So, there is a understanding that was quickly recognized that the channel that you actually are experiencing sometimes can be much better than what you need because of the margins that are present is that a reasonable statement because you have put in idea, this is a buffer in case you need it is there, but if you do if your channel conditions is good then you actually have more than what you need. So, this was a observation that was made very very early, but of course, there is no way you can get rid of this margin because then you will have a poor performance.

So, there was a interesting discussion that took place, and I take you back almost to the year 1990 where this discussion is happening, this discussion is about the Vocoder remember we talked about the vocoder being the digital representation of voice. Analog transmission basically oh sorry, analog transmission was using waveforms the a d p; p c m transmission was 64 kilo bits per second, now GSM has come along they need to pick a data rate for the vocoder.

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So, here is a piece of information which we will validate a of few minutes later, 1 GSM user when you are assigned a traffic channel you basically have a data rate of 22.8 kilo

bits per second, keep that as a number for now, we will fix it, we will clarify it in a few minutes.

Now, the vocoder has got a certain source compression then it has got a on top of that you will have to add channel coding. So, what we see on the right hand side are 3 modes just illustrative examples. Mode 1 which has got the best speech quality it takes this 22.8 kilo bits per second and says I am going to use 16 kilo bits for representing the speech 6.8 kilo bits per second for the error protection, priority for speech small amount of error protection.

Now this light blue which is mode 3 we can think of it as the case where you said I am going to use only 8 kilo bits per second for speech, and I am going to use 14.8 kilo bits per second for the error protection; that means, you are going to put more emphasis on error protection and as all good communication engineers know error protection is only overhead that you are adding to make sure that your information is reliably transferred, at the end of the day what will determine the quality of the speech at the other side is how much speech rate will you have. So, how much compression do, the more you compress the less the quality will be.

So, look at this particular plot it has a on the y axis against a qualitative plot, y axis is speech quality, on the right on the x axis it is your channel conditions signal to noise ratio or basically your carrier to noise ratio basically how much impairment that you are having. Now when you have a lot of impairment which mode what you want to use?

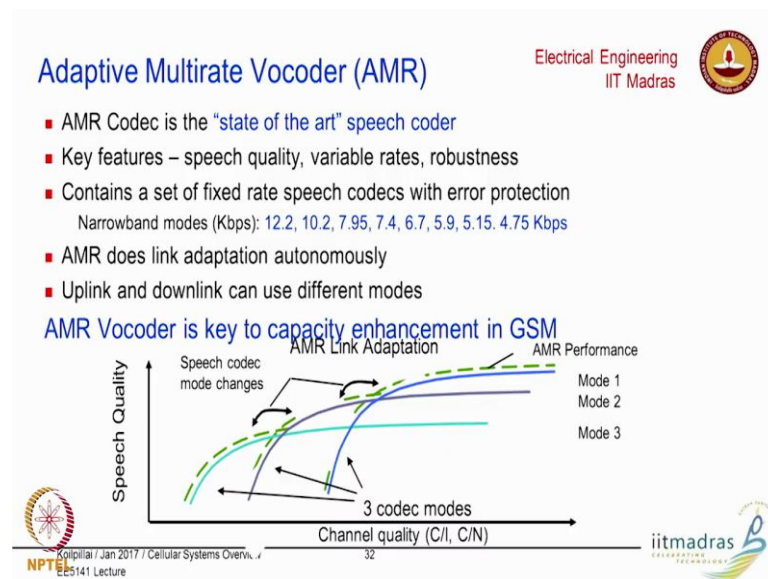
Student: 3.

3, because why because that has got the maximum error protection. So, you can see that mode 3 will survive even when the channel conditions are not so good, on the other hand mode what, but it will only give you a certain level of quality. On the other hand mode 1 will give you much better quality under good channel conditions, but notice that it will collapse much earlier than mode 3 and mode 2 is somewhere in between. So, the big debate was do we go for mode 1 or mode 2 or mode 3 and which mode do you think 1?

Student: Mode 3.

Mode 3 because at the end of the day they said look this is a highly dynamic channel we do not fully understand GSM is a brand new system, we do not want users to complaints saying you know my voice quality is not good. So, we want a robust system. So, at the end of the day they said we go with mode 3, but very soon everyone realize that you know a lot of times you are in channel conditions in which mode 1 could actually have been used. And therefore, the next version the enhancement of the cellular systems did this.

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They introduced something called adaptive vocoder which basically had a lot of data rates all the way from 4.75 to 12.2. Notice that you can go down to 4.75 which means that you can have a huge amount of error protection and the good thing is without user knowing the system will determine what the channel conditions are, and will automatically switch between the different modes. Robust when you need when your bad channel conditions and good channel conditions it will go to the, the 1 set have got higher speech coding rates. So, this was a concept that was seen and leveraged in the context of voice.

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Adaptive Modulation

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Adaptive Modulation on a burst by burst basis

- Each subscriber operates at the data rate corresponding to its link quality

Base Station

QPSK QAM 16 QAM 64

Sub #1 Sub #2 Sub #3 Sub #N

Ref: K. Kuchi (CEWiT) "Mobile WiMAX Tutorial"

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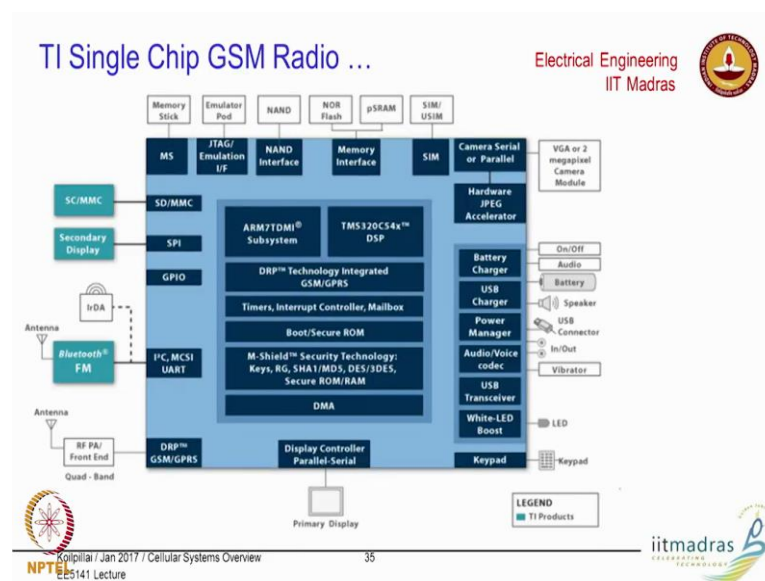
So, now when we come to packet communications the same principles apply. So, therefore, the understanding is we do not have a single modulation method it is going to be a dynamically varying modulation method where you could go from QPSK when the user is far away from you, which means that the user signal conditions are not so good, they are probably seeing a lot of interference from other base stations because they are far away from your own base station. So, this is probably user who will have a poorer channel conditions you will give the person only a lower modulation scheme.

Somebody in the middle may get slightly better 16 QAM, but then they may go into a building or shadowing effects immediately they will be downgraded to QPSK or if they start moving and they move towards the base station then you give them 16 QAM or even higher data rates. So, this notion of adaptation based on channel conditions has become the underlying foundation on which the third generation, fourth and fifth will be built.

So, the notion that it is a dynamic channel we have to build a system robust for the dynamic channel and in order to take advantage when the channel conditions are good we fully leverage our understanding of the and keep in mind that this good channel condition may not last very long you have to be able to switch and of course, you cannot expect the user to in to tell you when to switch to better or lower it all has to be done in a very dynamic fashion and this has become a part of the system that we have designed.

So, we are feeling very good because you know we have understood the wireless channel, we understood why they have all these fading margins, and we look now and say you know go to the handset manufacturers and say you know what without our understanding of the wireless channel you know your device will not cell, because you know people will not buy it because the channel conditions, but then you know our good feelings are do not last very long because they show us what is inside the phone. So, this is a GSM phone a designed by Texas instruments it is a single chip phone and I look for oh ok, things that you recognize.

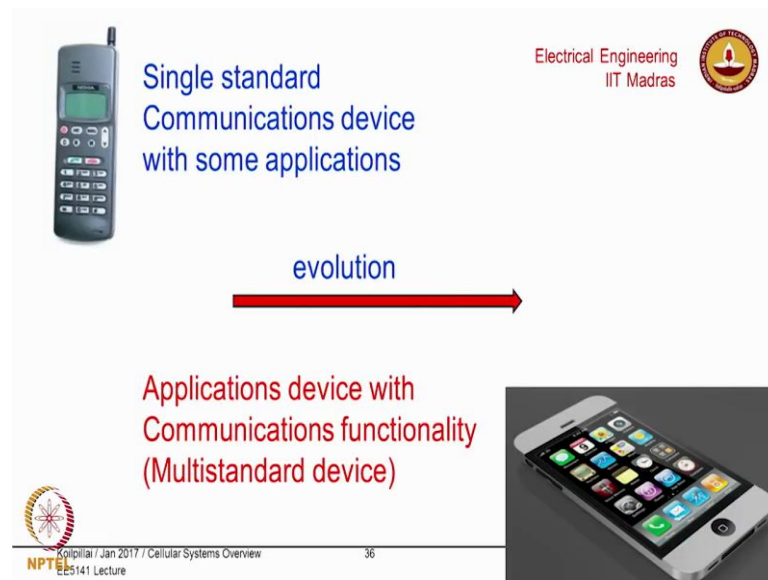
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Arm, microprocessor DSP, I see camera interface and I see all kinds of memory interfaces, I see you know LED display keypad and say where is GSM, somewhere in the corner is GSM, because that is how much silicon is needed for implementing the GSM functionality or the communications functionality.

What else is there? Bluetooth, infrared, basically the phone it has got lots of functions, but the master or the queen of functions is sitting in the left hand bottom corner that is what we are studying. So, without this the rest of it is not you are not very useful. So, a still there is a role for us to play, but we are a little bit more humble saying that yeah it is a small part, but a very important part.

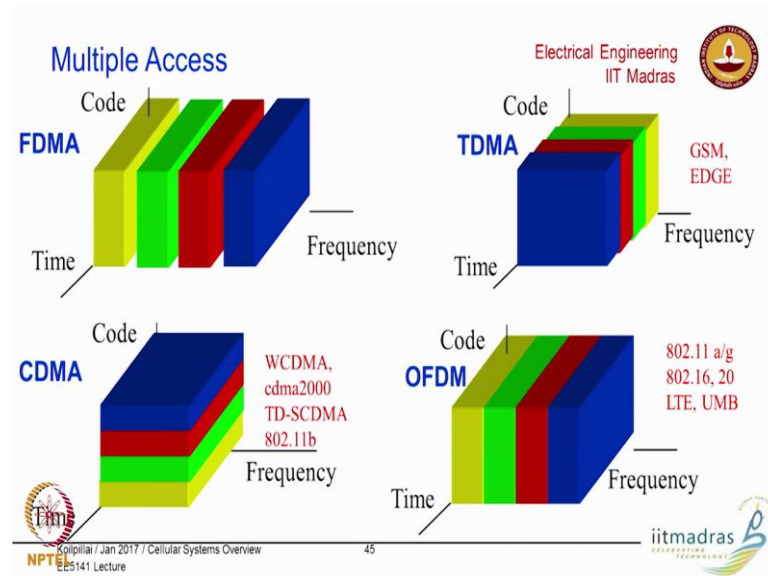
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So, what is been the evolution? It was once upon a time a single device with a single functionality which was voice, today it is got all kinds of functionalities, but voice happens to be 1 of them, but it is 1 where the RF connectivity is a key for us to be able to establish what we want to do. So, I want to quickly move into the different generations of phones or of a cellular systems, and spend a few minutes trying to remind you of what you may have already have studied in the digital communications.

So, multiple access, it is a shared medium wireless, so large number of users I have to keep them orthogonal to each other - how do I do that and what is the best way to establish orthogonality.

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So, first generation systems said that we will give each user different frequencies. So, in a cell each user got different frequency. So, think of it as 3 orthogonal axis and I will tell you how to visualize them; x axis being frequency, y axis being time, and z axis being code or power just think of it as a third dimension will become clearer as we move through this sequence of slides. So, notice that a user is given a unique frequency there is a separation between two users, therefore, they will not interfere with each other. And when it is when each user can transmit for as much time as they want there is no restriction they can transmit and they can transmit with full power maximum power because yellow transmitting with maximum power does not affect green or red each one is orthogonal in terms of frequency. So, this was the first generation systems FDMA base systems.

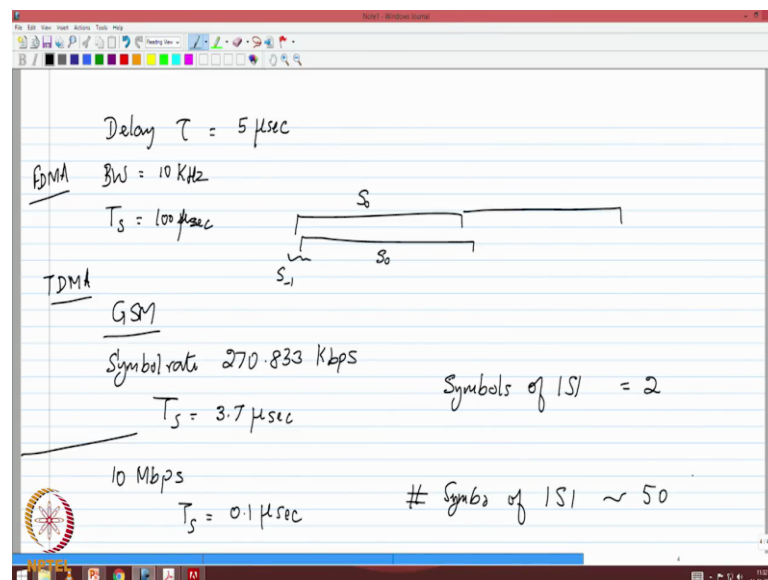
We found that there is a lot of wastage of resources because there is a guard bands between them, so second generation said we will do things a little bit differently. We will change the orthogonality from frequency to time. So, everybody has the same frequency, but we will now have time slots when it is your time slot you transmit you can transmit with maximum power, but at that time nobody else will be transmitting because it is not their turn to transmit.

So, basically you get a pattern of time slots and then the pattern repeats. So, you get a time slot in every frame. So, this turned out to be quite attractive because a time we can

get very precise clocks and therefore, designed the time division multiple access systems this was a very interesting way, this was the way we could design the systems. Now I want you to start thinking about another aspect which is very important for us and that aspect is the aspect of equalization.

So, let me just sort of switch to the screen for a moment, and ask you to think about the aspect of. So, let us say that the delay in the channel; that means, the time gap between the first and the last arriving paths is we call let us call that as tau is 5 microseconds, 5 microseconds.

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now when I did FDMA let us say that we gave each user 10 kilo hertz the bandwidth in FDMA, FDMA is the first generation system bandwidth is 10 kilo hertz; what is the symbol duration if you are using binary transmission? 100 microseconds. Now if I were to ask you how much inter symbol interference is going to be between adjacent symbols. So, if you had to think of one symbol like this the next one is going to arrive like this. So, the amount of inter symbol interference is very very minimal because the 5 microseconds in 100 microseconds the duration of the symbol is 100 microseconds the delay between the first and second path. So, it almost looks like there is no inter symbol interference would you agree anybody not clear about the symbol duration and delay spread how it impacts each other let us move to TDMA.

Student: Can you repeat a last part?

Let us say that the bandwidth of the signal that I am transmitting is 10 kilo hertz binary transmission. So, approximately one symbol, basically it is 10 kilo board is my signaling rate. So, basically if 10 kilo board is my signaling rate, my symbol duration is 100 microseconds basically were reciprocal of the of the baud rate. So, 100 microseconds will be 1 symbol and the next symbol will come after that the delayed version is going to come with a delay of 5 microseconds. So, basically the inter symbol interference is going to be caused by the fact that there is a there are two copies of the signal which are slightly overlapping with each other, but the bulk of the overlap is common basic the same symbol is coming on both branches there is a little bit of the previous one. So, if this is S 0, this is also S 0, this will be S of minus 1, there is a little bit of S minus 1 that is coming in and therefore, you have to make sure you take care of that, but you can almost choose to ignore it there you may not pay a penalty for it.

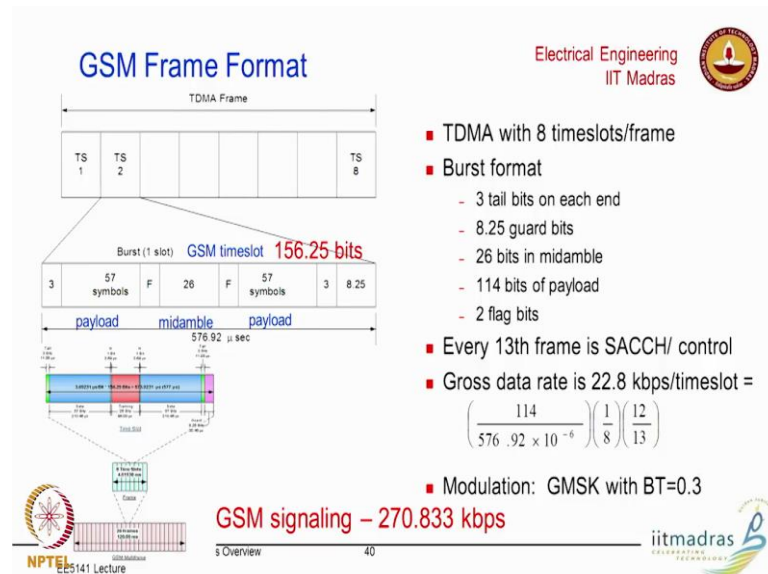
TDMA - let us take GSM, GSM of the baud rate the symbol rate, symbol rate is 270.833 kilo bits per second it is the binary modulation scheme. So, the symbol duration is a approximately 3.7 microseconds, how many symbols of inter symbol interference do I see? 3.7 is the symbol duration there is a delay of 5 microseconds and then a copy would come. So, how much is I would I see? 2 symbols. So, the number of symbols of ISI symbols of ISI it is two symbols, 2 to 3 depends on how the alignment occurs you may have to worry about a third symbol in your demodulation, but it is still manageable we can design a equalizers.

Now if I want to increase my data rate let us say I want to go to 10 megabits per second, this is using TDMA methods. So, I want to increase I want to have more users. So, therefore, FDMA means only 1 user at a time TDMA, GSM has got 8 users in a frame I want to increase the 10 megabits per second. Now if I try to do that notice my symbol duration what does it become 0.1 microseconds

How much of ISI will I see number of symbols of ISI? Number of symbols of ISI almost 50 it is going to be very very difficult for you to design an equalizer for 50 symbols of inter symbol is a very complex system. So, TDMA seems to have a limitation in that if I keep increasing the data rate and keep adding more and more time slots to support users or to get achieve high data rates, the place where I am going to pay a penalty is going to be in the complexity of the equalization yes I can maintain orthogonality in time, but the receiver is going to be more complex. So, therefore, I have to be careful with that. So, let

us keep that picture in mind. So, that was second generation we were happy with users being satisfied with a 100 kilobits per second, but third generation came along people said, no we want the data rates packet data should go to 14 megabits per second and higher. If you can go back and look at the slides third generation was targeting around 14 megabits per second.

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So, before we get to that let us remember we said 1 time slot of GSM gives you 22.8 kilobits per second as the raw payload. Let us do a quick calculation to get you familiar with the TDMA concept; at the bottom of the screen that is your GSM signaling rate 270.833 kilobits per second, a GSM frame is divided into 8 time slots and each time slot has got the following structure. There are two places where user data is sent that is basically the payload 1 and payload 2 each of them has got 57 symbols or 57 bits, in the middle that is overhead 26 bits for a for you to do the synchronization, there are flags which tell us whether this is user data or control information and then there are some guard bits on either side.

So, part of the bursts and then there are 3 plus 8 and a quarter at the end of the burst why 8 and a quarter I will tell you sometimes there is only some history to that, but basically this is the structure. A frame has got 114 bits of payload useful information and the rest is overhead of one kind or the other. If you add it up it will come to 156.25 bits that is approximately 576 microseconds if you look at it at this data rate.

Now, this frame of GSM is embedded in what is called a multi frame structure, the system counts not on the basis of individual frames on multi frames and the multi frame structure is based on a structure of 26 frames. 26 frames make a multi frame that is what you see at the bottom and that is 120 milliseconds that is a nice number. So, the clock keeps track of 120 milliseconds and so in this 26 frames each user should have got 26 time slots because each frame you should get 1 time slot. Now every 13th frame is used for control. So, basically out of 26 only 24 will be used for user data. Again it is if there is a numerology involved in it, but once somebody explains it should be fairly straightforward.

So, now what is the data rate per user - I get 114 bits of payload I get it once in 576 microseconds, so 114 bits every 576 microseconds that is my data rate. But I do not get it all the time I get it 1 out of 8 time slots. So, there is a factor of 1 over 8 because it is a TDMA system. And I also have to set aside 2 out of 26 frames for control information for doing sending some measurements. So, therefore, there is a factor of 24 by 26 or 12 by 13 as the factor that also at the end of the day this is what I as 1 user will get. 114 bits every frame 24 out of 26 frames 1 out of 8 time slots do the rough calculation it should come out to 22.8.

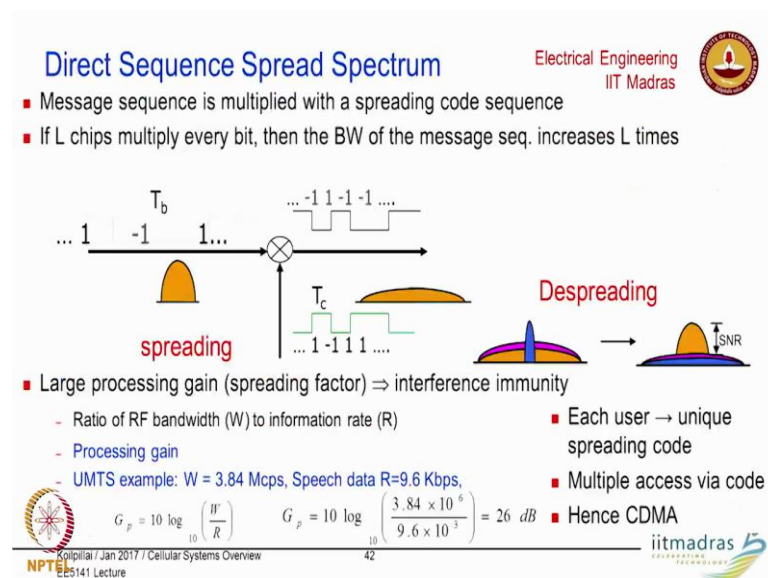
So, what was my signaling rate? 270.833, I divide it by 8 I should get some number much higher, but because of all the overheads what I eventually get is only 22.8 it is actually a fairly well designed system the amount of overhead is kept to a minimum it is a very very robust system, but this is how TDMA works. You have a raw signaling rate you got all kinds of overheads then you get what is the user frame and make sure that you do not forget this 1 over 8 times a factor because that is what tells you that you are not you do not want all the time slots.

Now as GSM has evolved when people wanted to have higher data rates what one of the options was? To give you more than 1 time slot. So, if you got 2 time slots then you could go 22.8 times 2 and then eventually you can ask for all 8 time slots that is a possibility. So, the way to get it is to ask for more time slots in a TDMA system. So, that is TDMA complexity will be in the equalization, GSM, well designed because you got only two to 3 symbols of ISI, but once you go to 10 megabits per second a TDMA system will have some difficulty. So, then came along a system which is based on spread spectrum technology. Spread spectrum technology says that orthogonality will not be in

frequency or in time it will be in the third dimension which is the code or a signature sequence these signature sequences are inherently orthogonal. So, therefore, they will not interfere with each other though you are using the same frequency and transmitting at the same time.

So, for example, all these 4 users can use the same frequency at the same time, but they have orthogonal signature sequences and therefore, they do not interfere with each other. So, the question that was asked yesterday is you know why not add more and more users I am keeping that you know this is a good looks like a very good way to increase. So, let us see where the limitations will occur.

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I am assuming most of you are familiar with the spread spectrum, but just as a concept diagram the input data is a narrowband data, I spread or basically multiply it with a spreading sequence in this case I have used a 4 bit spreading sequence. So, using a 4 bit spreading sequence I have basically multiplying it. So, what would have been 1 bit per second now becomes 4 chips per second these are called chips to differentiate between what is the information input rate and the channel rate. So, channel is measured in terms of chip rate. So, 1 bit per second becomes 4 chips per second. So, basically the bandwidth expands by a factor of 4. If you had done a factor of 16 chips per symbol it would have spread by a factor of 16. So, the controlling factor is how much spreading do you want to use.

So, the good thing is you can find several orthogonal sequences and therefore, start using different users orange pink you can keep adding those users, how does the receiver work? The receiver applies the same sequence basically multiplies the received signal by the same sequence as the spreading sequence. So, if you multiply 1 1 minus 1 minus 1 by 1 one minus 1 minus 1 minus 1 1 minus 1 it will get 4. So, basically it will recover the original signal if it matches if it does not match it will reject it. So, that is the underlying principle of a spread spectrum signal.

Now, what happens if there is a delay spread, supposing you shift by 1 bit these spreading sequences have got the property that even if you spread even if you are off by 1 chip they the correlation will be very low. So, inter symbol interference is sort of inherently suppressed in a spread spectrum system. So, you will reject other users you will also reject inter symbol interference again to a limit, but the important thing is to know is how much benefit are you getting. So, a simple example this is measured in terms of processing gain in the third generation system speech at 9.6 kilobits per second, I actually use it is through spreading it is taken up to 3.84 mega chips per second. So, the ratio of these two to a logarithm scale is 26 dB that is the processing gain; that means, that if there is a interferer whose 26 dB stronger than me I will still survive and still get my signal out. Because I have it is effectively like saying my signal will be amplified by 26 dB compared to everybody else.

So, that is the advantage each user is given a unique spreading code and multiple access is through the spreading codes and because you are using codes it is called code division multiple access and I am sure most of this is known material to you.

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CDMA

- Walsh-Hadamard Codes
- Used as Spreading Codes in CDMA systems
- Cdma2000 uses length 64 spreading codes

$$[1] \rightarrow \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \rightarrow \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix} \rightarrow \begin{bmatrix} 1 & 1 & 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 & 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 & 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 & 1 & -1 & -1 & 1 \\ 1 & 1 & 1 & 1 & -1 & -1 & -1 & -1 \\ 1 & -1 & 1 & -1 & -1 & 1 & -1 & 1 \\ 1 & 1 & -1 & -1 & -1 & 1 & 1 & 1 \\ 1 & -1 & -1 & 1 & -1 & 1 & 1 & -1 \end{bmatrix}$$

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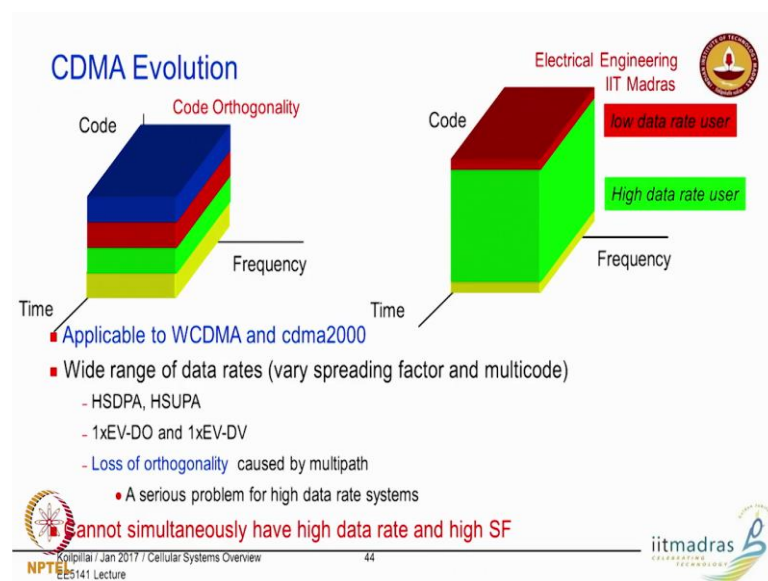
How do we generate codes? One of the simplest ways is the Walsh-Hadamard codes if I say that I want to have spreading factor of 4 how many orthogonal codes I have is only 4. If I go to 8 and then to 16 I will get whatever is the size of the matrix that is the number of go I can of course, go to quasi our orthogonal codes, but at some point I will run out of the spreading sequences. So, this is where the limit has been. So, CDMA 2000 if you had used a CDMA phone was using a spreading factor of 64. So, theoretically you could use 64 users in a particular cell you could give them all and they would not interfere with each other in practice it was found that you could use anywhere up to 30 to 32 user codes after that because of loss of orthogonality and I will explain in a minute where the loss of orthogonality occurs you started to see degradation in the performance.

So, basically you even if you had codes the performance of the system may limit the amount of spreading that you could do. So, now, comes a very interesting challenge 9.6 kilobits per second I can spread by you know 256 and still get a system, but if I want to transmit a 10 megabits per second, let us say I want a user says my fourth generation system is going to transmit a 10 megabits per second and you say well good I will design for you a spreading spread spectrum system factor of 64 I will spread to make it very robust to give 64 users each of them getting 10 megabits per second, how much bandwidth do I need? 10 megabits per second assume binary to each 10 megahertz is what I will need if I do not spread if I spread by a factor of 64; 64 megahertz and how

much operator all our operators put together do not have 64 megahertz leave alone 11 user.

So, and then you say well you know for broadband systems maybe CDMA has got a problem, because I do not have the bandwidth well somebody was very smart they said you know what I am going to reduce the spreading factor you know when you ask low data rate I will give you 64 spreading when you ask high data rate I will give you only 4 spreading factor.

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So, when you use less spreading factor; that means, you do not have much processing gain. So, where should you compensate by transmitting more power. So, what do you do? This is a scenario where everybody is using the same data rate, here is the case where a CDMA system where 1 user is using low data rate and the green user is asking for high data rate and because I do not want to expand my bandwidth I am reducing a spreading factor.

So, therefore, I have to increase this power; looks very nice except if there is loss of orthogonality. Loss of orthogonality means these colors mix with each other, and then what happens the green user will wipe out everybody else because he is transmitting at very high power. So, that is the problem that we see in the last line it is very difficult to have high data rate and high spreading factor you have to reduce the spreading factor which means that you are going to transmit with high power which means that the

minute orthogonality is lost there for you. So, when is orthogonality lost when there is multi path, now in a wireless system I can never guarantee you a channel where there is no multi path. So, therefore, spreading factors with CDMA there is a challenge.

So, the back to the drawing board I want broadband systems I do not want the complexity of equalization I do not have a bandwidth to spread what do I do. So, we went a full circle and said well you know FDMA is probably the most robust system provided we can get rid of those guard bands, we get rid of the guard bands it is very good because you know each of these only 10 kilohertz ISI negligible, I can create as much bandwidth as I want without increasing complexity, there is no limit and I can just keep adding more and more of these narrowband carriers and that happens to be OFDM orthogonal frequency division multiplexing. This is, these mode by which we have built the fourth generation very likely some of the fifth generation will also be some variant of this area of the OFDM system.


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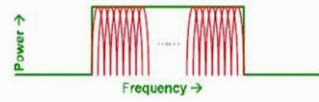
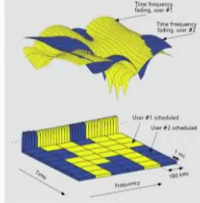
OFDM


- Effect of Multipath
 - CDMA – loss of orthogonality
 - More severe if spreading factor is low
 - TDMA \Rightarrow need for complex equalization
 - More severe for higher baud rates
 - OFDM attractive for high speed data in multipath fading
- OFDM – Orthogonal Freq Division Multiplexing (Multicarrier)
 - Narrow carriers \Rightarrow low baud rate \Rightarrow long symbol duration
 - An attractive candidate for **broadband wireless**
 - Efficient digital multicarrier implementation using DFT/IDFT
 - Opportunity to do optimized coding and modulation in each carrier
 - Maximize capacity utilization based on channel condition
 - A active area of research
 - Issues: High peak-to-average ratio, sensitivity to frequency & timing errors
- OFDM used for WLAN, WWAN, Digital Audio Broadcasting, 4G, ...

OFDM \Leftrightarrow Multi-carrier Modulation \Leftrightarrow Multi-tone Modulation

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So, to summarize CDMA if I lose the orthogonality I have a problem I cannot of course, spread for high data rate systems. So, therefore, I have the issue of orthogonality which is caused by multi path. For TDMA systems the complexity of equalization becomes a factor OFDM is very attractive because I do not need to do equalization most of the time because my carriers are all narrowband, my symbols in each carrier is quite long. So, therefore, I deal with it on a carrier by carrier basis. In addition to that OFDM I am sure

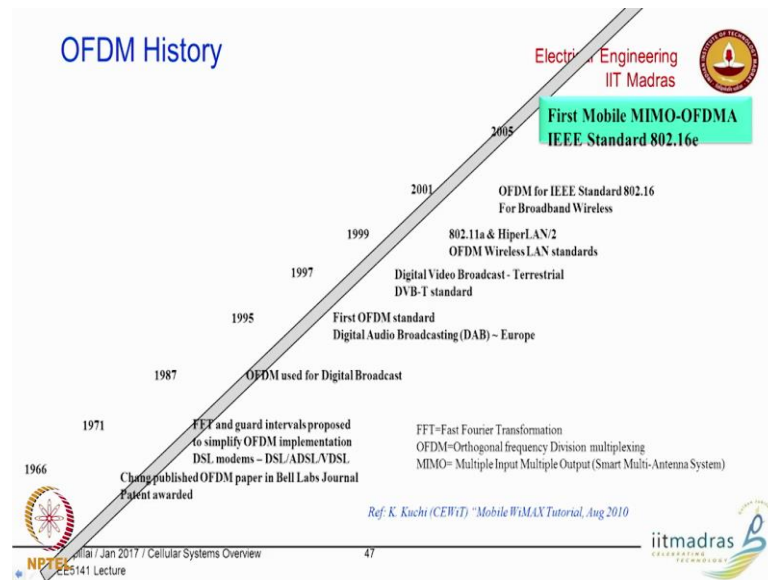
if you have studied it or we will be doing it towards the end of the course is built on the IDFT and the DFT discrete Fourier transform very efficient implementation.

So, therefore, it is a system that is very attractive from the point of view of the implementation. So, then everybody should go to AFDM right everybody why should throw out TDMA, FDMA, CDMA and then everybody should go to OFDM. So, there must be a catch somewhere there must be some price that you have to pay it turns out to be that there is this parameter called peak to average ratio, which is the $PAPR$ which is going to determine the efficiency of your power amplifier and that is not very good for an OFDM system. In fact, it is much higher than a CDMA system or a TDMA system and that is where the price we pay.

You will have to generate more power to transmit at the same level as the other systems. If you have to generate more power in your handset what is happening your basically if your power if amplifier is operating at lower efficiency it is being generated as heat, and if it is heat well tell a guess where who is absorbing the heat it is you because if it is next to your face. So, basically a peak to average ratio is an important one it is a price that we pay for OFDM, but it is got lots of advantages. So, the way to think about it is yes take the advantages find a solution for the peak average power ratio challenge and that is what we have seen in. Another very very interesting phenomenon or aspect about OFDM, think of this figure in the lower part of the figure you have a large number of small carriers narrowband carriers.

Let us say I have two users, and these two users are very dynamic they are both are accessing the internet you know suddenly one person is asking for high data rate and then is then idle for some time then in the middle some the other person is asking. So, dynamically they are asking for different data resource requirements. So, see how if this is your time axis at different time instances you can allocate different number of subcarriers to each user. So, basically you dynamically change the allocation based on their current requirements not only that remember the fading phenomenon if the yellow user has got some channels which are affected by fades you can give him some other channels which are not affected by fade. So, again you can overcome the fading aspect you can over you can you can address the dynamic need requirements and you can also overcome the equalization as issues.

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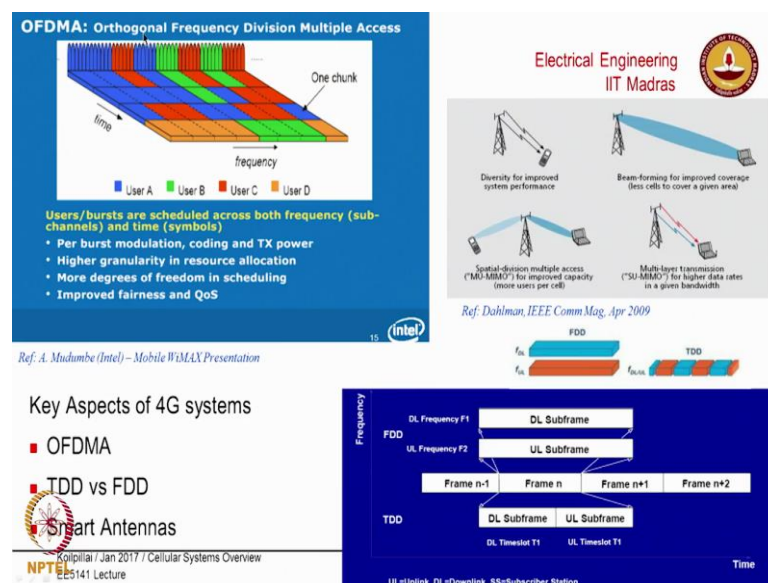
So, again OFDM has got some very very attractive advantages. Now you may say who discovered OFDM wireless guys discovered OFDM right fourth generation, the answer is no you know 50 years almost 30 years before wireless was invented they had already come up with the OFDM concept they did not call it OFDM they called it multi tone transmission multi tone meaning multiple carriers, it was a different terminology. It was invented for by a telephone company the wire line telephone company basically bell labs. So, the notion was a wire line channel copper has got a large bandwidth it got about 4 5 megahertz of bandwidth, how much bandwidth do we use for transmitting voice 4 kilohertz it is a huge waste of resources.

What happens to all the other bandwidth that you are using unfortunately the gain is very very poor in some of the as you go to high frequencies. So, the notion of using multi tones transmitting at different power levels, to exploit the extra bandwidth that was there on the copper channels. So, you see why it was originally designed to take advantage of channels which did not have good gain, but still get capacity out of it and if you go back and look at all the modems that have been used multi tone modems, you will find it under the family called DSL family digital subscriber loop; basically it is data rate over copper wires. So, initially it was advanced a digital subscriber loop, very high data rates as asymmetric digital subscriber loop very high data rate subscribe a loop now they just call it ex DSL.

So, basically very high bandwidths you can get over copper, if you have good quality copper unfortunately India does not have good quality copper, we do not have much copper to begin with so, but if you have good quality copper you can get very high data rates several megabits per second.

Now that has been the starting point of OFDM, but now the wireless people have adopted it for the several advantages that it gives us and therefore, we continued development has been taken up by the wireless community. So, let me just give you introduction to 4 G systems and then we will stop there. So, if I were to ask in a nutshell, I have understood multi multiple access, I have understood you know what the earliest generations did what were some of the factors that enabled higher data rates to be achieved in the fourth generation systems.

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So, first was OFDM that I could divide multiple carriers and then give allocate to different users and use it very dynamically that was one. The second aspect was the introduction of multiple antennas, if you remember I mentioned antennas can give you diversity they can also give you capacity. Let me just give you quickly this diagram on the top right hand corner.

Diversity means using both antennas to transmit the same signal two copies of the signal; multiplexing means you transmit different copies of the signal to the same user so that means, this guy gets double the data rate, but you still want to take advantage of the fact

that you have two antennas and therefore, all the advantages of a MIMO system multiple antennas at the transmitter, multiple antennas at the receiver are being exploited in a 4 G system.

So, if you have multiple antennas at the transmitter you can do beam forming only radiate in the direction of the user. So, you can have more range you can also radiate beams to different users. So, which means that you can have support more users or to the same user you can get more data rate that is all the advantages. Third aspect of fourth generation systems is that traditionally a wireless systems have been frequency division duplex systems, uplink the way you communicate to the base station is at a high different frequency at the base station communicates to you, but in the fourth generation we have also introduced the time division duplex where you have only one channel, part of the time you communicate to the base station the rest of the time the base station communicates to you. So, basically we have a downlink uplink and it goes ping pong methods.

So, there are some advantages with this there are, so FDD means frequency division duplex you have different frequencies type TDD means that you are doing on the same frequency uplink and downlink are happening. So, these are some key elements of the fourth generation. There are some more elements of fourth generation which are based on the technologies that have evolved around cellular technology; cellular systems and I will pick it up from there in the next lecture. But please do read I think at this point you should be able to cover a lot of the slides which are starting from here going on. The last part we will talk a little bit about the first generation systems which will be completed in the next lecture.

Thank you, we will see you tomorrow.