Introduction to various generations of mobile phone technologies.

The first generation mobile phones :

The first generation was designed for voice communication .one e.g is AMPS- Advanced mobile phone systems, used in North America in Bell labs.AMPS is lledan analog cellular phone system.A geographic region is divided up into cells which is why devices are sometimes called cell phones.in AMPS ,the cells are typically 10 to 20 km across,in digital systems,the cells are smaller.each cell uses some set of frequencies not used by any of its neighbours. the idea that gives cellular systems far more capacity than previous systems is the use of small cells and the reuse of frequencies in the nearby cells(not the adjacent cells).An IMTS(Improved mobile telephone systems)100 km across can have one call in each frequency ,an AMPS system might have 100 10-km cells in the same area and be able to have 10to 15 calls on each frequency in widely separated cells.Furthurmore,smaller cells mean that less power is needed,which leads to smaller and cheaper transmitters and handsets.

Channels:

The AMPS system uses 832 full duplex channels,each consisting of a pair of simplex channels. There are 832 simplex transmission channels from 824 to 849 MHz and 832 simplex receive channels from 869 to 894 MHz.each of these simplex channels is 30khz wide.Thus,AMPS uses FDM to separate t he channels. In the 800-MHz band,radio waves are about 40 cm long and travel in straight lines.they are absorbed by trees and plants and bounce off the ground and buildings. it is possible that a signal sent by a mobile telephone will reach the base station by the direct path, but also slightly later after bouncing off the ground or a building .this may lead to an echo or signal distortion(multipath fading) .Sometime,it is even possible to hear a distant conversation that has bounced several times.

The 832 channels are divided into four categories:
1. Control(base to mobile) to manage the system.

2.Paging(base to mobile)to alert mobile users for them.

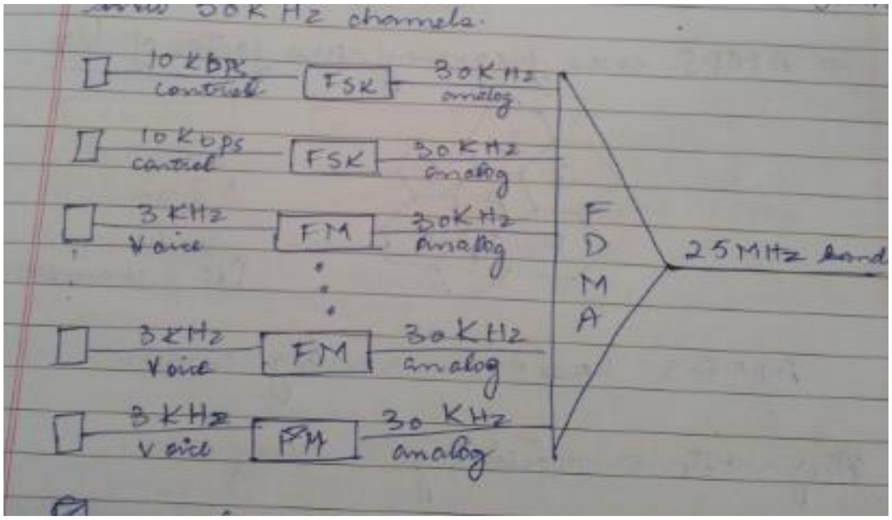
3.Access(bidirectional)for call setup and channel assignment.

4.Data(bidirectional) for voice,fax,or data.

Twenty –one of the channels are reserved for control,and these are wired into a PROM in each telephone. Since the same frequencies cannot be reused in nearby cells, the actual number of voice channels available per cell is much smaller than 832, typically about 45.

It uses 800 MHz ISM band and 2 separate analog channels; forward & reverse analog channels.The band between 824 to 849 MHz to used for reverse communication from Mobile Station to Base Station.The band between 869 to 894 MHz is used for forward communication from Base Station to Mobile Station. Forward channels divided into 832 channels of 30 KHz each. As each location area is shared by two service providers ,each provider can have 416 channels,out of which 21 are which are used for control

& remaining are used for voice communication. it uses FDMA to divide each 25-MHz band into 30KHz channels.



Here 3khz voice channel is frequency modulated to generate of analog channel. Out of 832 there are 790 voice channels are there where 3khz are frequency modulated to 30 khz.they are transmitted in parallel from mobile station to the receiver in base station.similarly forward channels do the same from transmitter of 25 MHz. Therefore 2 way communication takes place and two mobile stations communicate using one base station.AMPS uses frequency reuse factor 0f 1/7. Therefore number of channels is 832/7. AMPS based on analog communication.

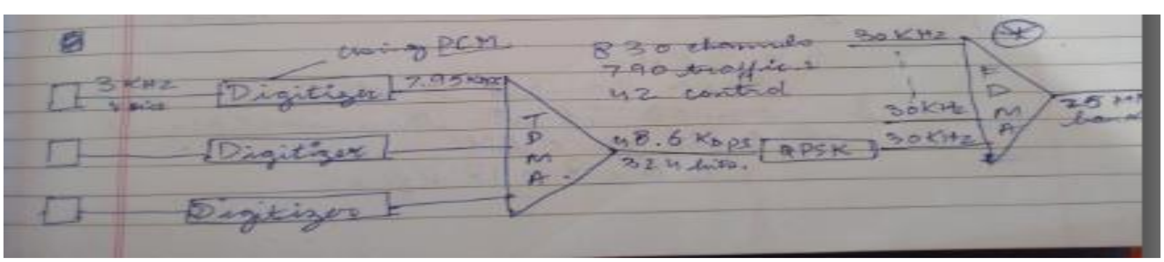
<u>2ND GENERATION</u>

Developed to provide higher quality mobile voice communication. Its designed for digitized voice.

1. D-AMPS(IS-136):uses TDMA and FDMA, digital AMPS,its an interstandard.
2. GSM:uses TDMA and FDMA, Developed in Europe.
3. CDMA(IS-95):uses CDMA and FDMA.

<u>D-AMPS</u>

Digital version of analog AMPS. Uses the same bands and channels as in AMPS.It has frequency reuse factor of 1/7.



Here 25 frames per second each of 1994 bits,is divided in 6 slots shared by three channels. Each slot has 324 bits—159 data bits,64 control bits,101 error correction bits.

If somebody is having analog handset and digital handset , they can communicate. They have to use the same bands and channels used by AMPS. It also uses the same frequency reuse factor. it is digital ,rest is same.

<u>How it is made digital?</u>

Here the 3KHz voice is digitized using a very complex PCM & compression technique to generate using 7.95 kbps .if we perform simple quantization, it will be at least 3*3 khz.they sample it at 6hz and then multiply it at 6 bit or 8 bit.i.e 6*8=48 kbps.But by use of certain compression technique, the digital signal is having 7.95 kbps.

So 3 such digital channels go through TDMA . In Time division ,to generate each of them is used.

Frames are being send and each of 1994 bits divided into six slots. 25 such frames per second each of 1994 bits and each slot is having 320 bits out of which data is only 159 bits,64 comtrol and 101 are used for error correction.

In this way we have got 48.6k bits and that 48.6 kilobits is converted into analog signal by using QPSK to generate 30 khz analog signal and that is being broadcasted and sent from the mobile station to the transmitter receiver of base station or from the base station to the mobile station.

<u>GSM-- THE GLOBAL SYSTEM FOR MOBILE COMMUNICATION</u>

The global system for mobile (GSM) communication is a european standard developed to replace the first generation technology.it uses two bands for duplex communication, but the frequency bands are different. For FB -935-960; each of 200 khz. For Mobile station to Base station we have 890-915, 124 channels each 200 khz.

Each voice channel is digitized and compressed to a 13 kbps digital signal.each slot carries 156.25 bits,8 slots are multiplexed together creating a FDM frame, 26 frames are combined to form a multiframe. 8 slots form a frame and 26 such frames form a multiframe.ds is repeated in 120 ms. If we combine all together 1/120(ms)*26(frame)*8(slots)*165.25(req msec)=270.8 kbps digital data. This digital data is converted into analog data by using GMSK(modified version of FSK)technique which sends it to the base station.Same happens for base station to mobile station,except that the frequency bands are different.it has 124 channels each can support 8 different users.GSM combines both TDMA & FDMA.Th ere is a large amount of overhead inTDMA, 114 bits are generated by adding extra bits for error correction. Because of complex error correction it allows a reuse factor as low as 1/3. i.e here R.F=1/3 and each cluster will be having 3 cells.

GSM -Reference architecture and components of mobile networks;their functions and characteristics. Use of HLR and VLR in mobile networks.

It consists of three major interconnected subsystems that interact between themselves and with the users through certain network interfaces.The subsystems are the Base Station Subsystem(BSS), Network and switching subsystem(NSS),and the operation support subsystem(OSS).The Mobile station (MS) is also a subsystem,but is usually considered to be part of the BSS for architecture purposes. Equipment and services are designed within GSM to support one or more of these specific subsystems. The BSS also known as radio subsystem, provides and manages radio transmission paths between the mobile stations and the mobile switching centre(MSC).The BSS also manages the radio interface between the mobile stations and all other subsystems of GSM. Each BSS consists of many BSCs which connect the MS to the NSS via the MSCs. The NSS manages the switching functions of the system and allows the MSCs to communicate with other networks such as the PSTN and ISDN.The OSS supports the operation and maintenance of GSM and allows system engineers to monitor,diagnose,and troubleshoot all aspects of the GSM system.it interacts with other GSM subsystems,and is provided solely for the staff of the GSM operating company which provides service facilities for the network.The MSs communicate with the BSS over the radio air interface.The BSS consists of many BSCs which connect to a single MSC,and each BSC typically controls up to several hundred Base Transceiver Stations(BTSs). Some of the BTSs may be co-located at the BSC, and others may be remotely distributed and physically connected to the BSC by microwave link or dedicated leased lines.Mobile Handoffs (called handovers,or HO,in the GSM specification)between two BTSs under the control of the same BSC are handled by the BSC,and not the MSC.This greatly reduces the switching burden of the MSC.

The interface which connects a BTS to a BSC is called the Abis interface.The Abis interface carries traffic and maintenance data, and is specified by GSM to be standardized for all manufacturers. In practice, however , the Abis for each GSM base station manufacturer has subtle differences, thereby forcing service provider to use the same manufacturer for the BTS and BSC equipment.

The BSCs are physically connected via dedicated lines or microwave link to the MSC.The interface between a BSC and a MSC is called the A interface, which is standardized within GSM.The A Interface uses an SS7 protocol called the Signaling Correction Control Part(SCCP)which supports communication between the MSC and the BSS,as well as network messages between the individual subscribers and the MSC. A Interface allows a service provider to use base stations and switching equipment made by different manufacturers.The NSS handles the switching of GSM calls between external networks and the BSCs in the radio subsystem and is also responsible for managing and providing external access to several customer databases.The MSC in the central unit in the NSS and controls the traffic among all of the BSCs. in the NSS there are three different databases called the Home location Register(HLR),Visitor Locator register(VLR),and the Authentication Centre(AUC). The HLR is the database which contains subscriber information and location information for each who resides in the same city as MSC. Each subscriber in a particular GSM market is assigned a unique International Mobile Subscriber Identity (IMSI) ,and this number is used to identify each home user. The VLR is a database which temporarily stores the IMSI and customer information for each roaming subscriber who is visiting the coverage area of a particular MSC. The VLR is linked between several adjoining MSCs in a particular market or geographic region and contains subscription information of every visiting user in the area.Once a roaming mobile is logged in the VLR, the MSC sends the necessary information to the visiting subscriber's HLR so that calls to the roaming mobile can be appropriately routed over the PSTN by the roaming user's HLR.The Authentication center is a strongly protected database which handles the authentication and encryption keys for every single subscriber in the HLR and VLR.The authentication center contains a register called the Equipment Identity Register(EIR)which identifies stolen or fraudulently altered phones that transmit identity data that does not match with information contained in either the HLR or VLR.

The OSS supports one or several Operation Maintenance Centers(OMC) which are used to monitor and maintain the performance of each MS,BS,BSC and MSC within a gsm system.The OSS has three main functions, which are

1)to maintain all telecommunications hardware and network operations with a particular market.

2)manage all charging and billing procedures

3)manage all mobile equipment in the system.

Within each GSM system,an OMC is dedicated to each of these tasks and has provisions for adjusting all base station parameters and billing procedures,as well as for providing system operators with the ability to determine the performance and integrity of each piece of subscriber equipment in the system.

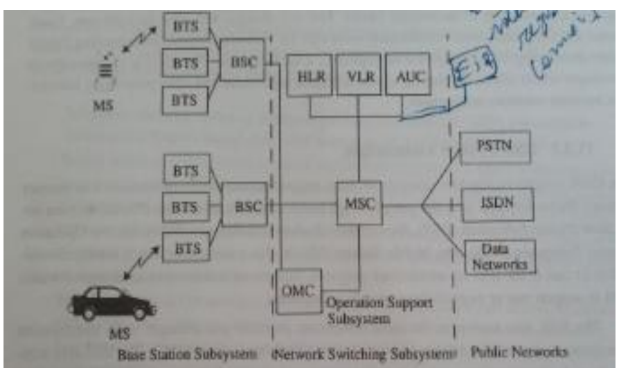


Figure 11.5 GSM system architecture.

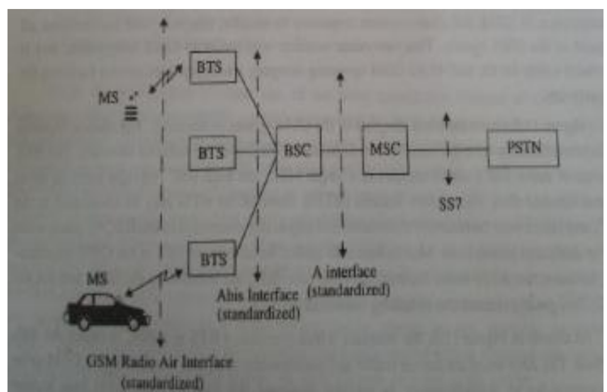


Figure 11.6 The various interfaces used in GSM.

HANDOFF SCENARIOS IN GSM

Handoff in the united states is referred to as handover in Europe and hence in GSM.The procedures that deals with mobility management.

INTERNAL HANDOVER: There are two types of Handover- internal and external.Internal handover is between BTSs that belong to the same BSS,and external handovers are between two different BSSs belonging to the same MSC. Sometimes there are handoffs betweenBSSs that are controlled by two different MSCs. In such a case, the old MSC continues to handle call management.Roaming between two MSCs,in inifiation of handoff two different countries is prohibited,and the call simply drops.

Handoff is initiated because of a variety of reasons. signal deterioration is the most common cause. Other reasons include traffic balancing where the handoff is network oriented to ease traffic congestion by moving calls in a highly congested cell to a lightly loaded cell.The handoff could be synchronous where the two calls involved are synchronized or it may be asynchronous. Because the MS does not have to resynchronize itself in the former scenario,the handoff delay is much smaller(100ms against 200 ms in the asynchronous case).

HANDOFF PROCEDURES INVOLVING A SINGLE MSC AND TWO BSSs

The BTS provides the MS with a list of available channels in neighbouring cells via the BCCH.The MS monitors the RSS from the BCCHs of these neighbouring cells and reports these values to the MSC using the SACCH.This is called mobile- assisted handoff. The BTS also monitors the RSS from the MS to make a handoff decision.Proprietary algorithms are used to decide when a handoff should be initiated.If a decision to make a handoff is made,the MSC negotiates a new channel with the new BSS and indicates to the MS that a handoff should be made using a handoff command.upon completion of the handoff,the MS indicates this with a handoff complete message to the MSC.

IS-95 CDMA

CDMA is quite complex in the sense the forward transmission and reverse transmission is different. Forward transmission is from the base station to the mobile stations.this transmission uses one type of signaling technique on the other hand mobile stations to the base stations uses a different type of signalling techniques.

Forward transmission

The CDMA forward channel is between the base station and the mobile station. The forward link in IS-95 occupies the same frequency spectrum as AMPS and IS-136 North American TDMA standards. Each carrier of the IS-95 occupies a 1.25MHz of band,whereas carriers of AMPS and IS- 136 each occupy 30khz of bandwidth. it consists of four types of logical channels-pilot channel, synchronization channel, paging channel, and traffic channels.Each carrier contains a pilot,a synchronization,upto seven paging and a number of traffic channels. These channel are separated from one another using different spreading codes.The modulation scheme employed for transmission of spread signal in the forward channel is QPSK.

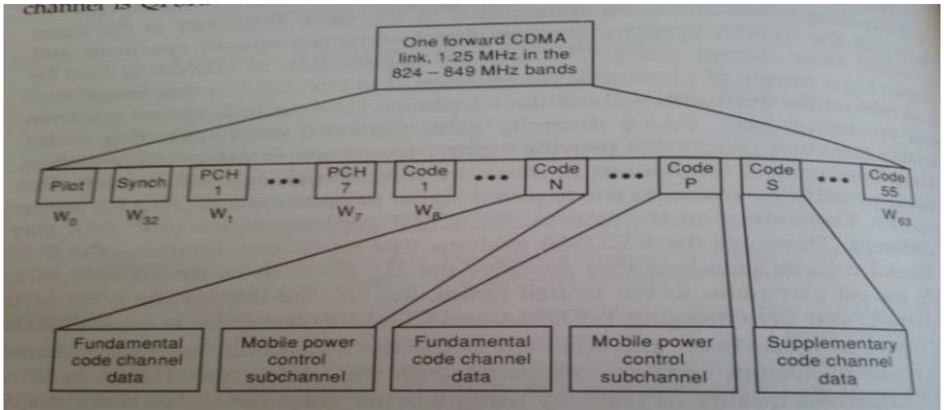


Fig. 9.16. IS-95 forward channels

Any information contained in the form of symbols is modulated by Walsh Codes which are obtained from Hadamard Matrices. Each walsh code identifies one of the 64 forward channels .after the channel symbols are spread using the orthogonal codes ,they are furthur scrambled in the in-phase and quadrature phase lines by what are called the short PN- spreading codes .they are not orthogonal but possess excellent auto-correlation and cross- correlation properties to minimize interference among different channels.They are M-Sequences generated by LFSRs of length 15 with a period of 32.768 chips.The orthogonal codes are used to isolate the transmissions between different channels within a cell, and the PN spreading codes are used to separate the transmission between different cells.the same PN sequence is used in all BSs ,but the PN sequence of each BS is offset from those of other BSs by same value. For this reason ,BSs in IS-95 have to be synchronizedon the downlink.such synchronization is achieved using GPS.

IS-95 REVERSE CHANNEL

Its fundamentally different from the forward channel.it employs OQPSK rather than QPSK . the OQPSK Iis closer to a constant envelope modulation,they provide for a more power efficient implementation of the transmitter at the MS. The QPSK modulation is easier for demodulation again at the MS. Compared with the forward channel,there is no spreading of the data symbols using orthogonal codes. Instead,the orthogonal codes are used for waveform encoding.This means that the reverse link employs an orthogonal modulation scheme that consumes bandwidth but reduces the error rate performance of the system.

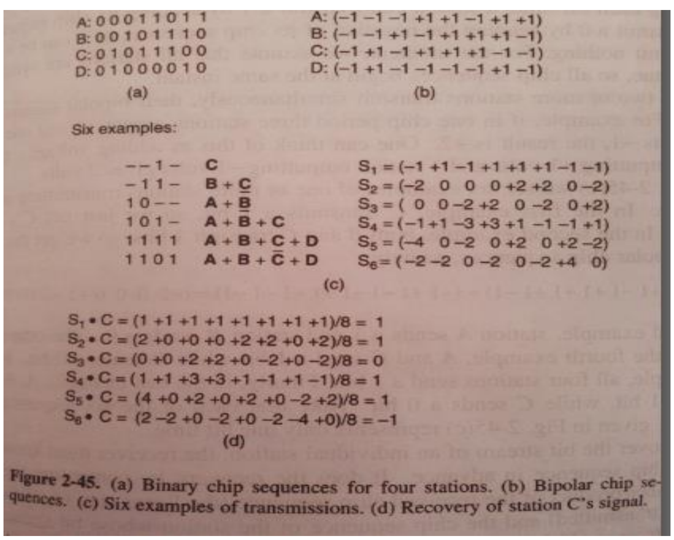
There are basically two types of reverse channels-access channels and reverse traffic channels.The reverse traffic channel,like the forward traffic channel,supports voice data at two rate sets-RS1 and RS2.In either case the data burst after coding and interleaving,but just before the 64-ary orthogonal modulation is at a rate of 28.8kbps.the output of the orthogonal modulator is 28.8 *64/6=307.2 kps. After spreading by the long PN code by a factor of four , the final chip rate is 307.2*4=1.2288 Mcps. A data randomizer is used in the fundamental code channel to mask out redundant data in case of symbol repetition .the reverse traffic channel sends the information related to the signal strength of the pilot and frame error rate statistics to the BS. It is also used to transmit control information to the BS such as a handoff completion message and a parameter response message.

<u>CDMA using chip sequences and concept and derivation of walsh codes.</u>

In CDMA, each bit time is subdivided into ‘m’ short intervals called chips.Typically, there are 64 or 128 chips per bit, but in the example given below we will use 8 chips/bit for simplicity. Each station is assigned a unique m-bit code called a chip sequence. To transmit a 1 bit,a station sends its chip sequence.To transmit a 0 bit, it sends the one’s compliment of its chip sequence.No other pattern are permitted.Thus, for m=8,if station A is assigned the chip sequence. No other patterns are permitted. Thus for m=8,if station A is assigned the chip sequence 00011011,it sends a 1 bit by sending 00011011 and a 0 bit by sending 11100100.

Increasing the amount of information to be send from b bits/sec to mb chips/sec can only be done if the Bandwidth available is increased by a factor of m,making CDMA a form of spread spectrum communication(assuming no changes in the modulation or encoding techniques).if we have a 1MHZ band available for 100 stations,with FDM each one would have 10khz and could send at 10kbps (assuming 1 bit per Hz).With CDMA each station uses the full 1 MHz,so the chip rate is 1 megachip per second. With fewer than 100 chips per bit, the effective bandwidth per station is higher for CDMA than FDM, and the channel allocation problem is also solved.

For pedagogical purposes,it is more convenient to use a bipolar notation, with binary 0 being -1 and binary 1 being +1. We will show chip sequences in parentheses, so a 1 bit for station A now becomes (-1-1-1+1+1-1+1+1). In fig (a) we show the binary chip sequences assigned to four example stations. (b)we show them in our bipolar notation.



A: 0 0 0 1 1 0 1 1
B: 0 0 1 0 1 1 1 0
C: 0 1 0 1 1 1 0 0
D: 0 1 0 0 0 0 1 0

(a)

A: (−1 −1 −1 +1 +1 −1 +1 +1)
B: (−1 −1 +1 −1 +1 +1 +1 −1)
C: (−1 +1 −1 +1 +1 +1 −1 −1)
D: (−1 +1 −1 −1 −1 −1 +1 −1)

(b)

Six examples:

– – 1 –	C	$S_1 = (-1\ +1\ -1\ +1\ +1\ +1\ -1\ -1)$
– 1 1 –	B + C	$S_2 = (-2\ \ 0\ \ 0\ \ 0\ +2\ +2\ \ 0\ -2)$
1 0 – –	$A + \overline{B}$	$S_3 = (\ 0\ \ 0\ -2\ +2\ \ 0\ -2\ \ 0\ +2)$
1 0 1 –	$A + \overline{B} + C$	$S_4 = (-1\ +1\ -3\ +3\ +1\ -1\ -1\ +1)$
1 1 1 1	A + B + C + D	$S_5 = (-4\ \ 0\ -2\ \ 0\ +2\ \ 0\ +2\ -2)$
1 1 0 1	$A + B + \overline{C} + D$	$S_6 = (-2\ -2\ \ 0\ -2\ \ 0\ -2\ +4\ \ 0)$

(c)

$S_1 \bullet C = (1\ +1\ +1\ +1\ +1\ +1\ +1\ +1)/8 = 1$
$S_2 \bullet C = (2\ +0\ +0\ +0\ +2\ +2\ +0\ +2)/8 = 1$
$S_3 \bullet C = (0\ +0\ +2\ +2\ +0\ -2\ +0\ -2)/8 = 0$
$S_4 \bullet C = (1\ +1\ +3\ +3\ +1\ -1\ +1\ -1)/8 = 1$
$S_5 \bullet C = (4\ +0\ +2\ +0\ +2\ +0\ -2\ +2)/8 = 1$
$S_6 \bullet C = (2\ -2\ +0\ -2\ +0\ -2\ -4\ +0)/8 = -1$

(d)

Figure 2-45. (a) Binary chip sequences for four stations. (b) Bipolar chip sequences. (c) Six examples of transmissions. (d) Recovery of station C’s signal.

Each station has its own unique chip sequence.Let us use the symbol **S** to indicate the m-chip vector for station S, and **S** NEG for its negation. All chip sequences are pairwise orthogonal, by which we mean that the normalized inner product of any two distinct chip sequences,(**S,T** written as **S.T**), is 0. It is known how to generate such orthogonal chip sequences using a method known as **Walsh Codes.** In mathematical terms, orthogonality of the chip sequences can be expressed as follows

$$\mathbf{S} \bullet \mathbf{T} \equiv \frac{1}{m} \sum_{i=1}^{m} S_i T_i = 0$$

As many pairs are the same as are different. Note that if **S.T =0,** then **S.T neg** is also 0. The normalized inner product of any chip sequence with itself is 1:

$$\mathbf{S} \bullet \mathbf{S} = \frac{1}{m} \sum_{i=1}^{m} S_i S_i = \frac{1}{m} \sum_{i=1}^{m} S_i^2 = \frac{1}{m} \sum_{i=1}^{m} (\pm 1)^2 = 1$$

This follows because each of the m terms in the inner product is 1, so the sum is m. Also note that **S.S NEG= -1.** During each bit time, a station can transmit a 1 by sending its chip sequence, it can transmit a 0 by sending the negative of its chip sequence, or it can be silent and transmit nothing. we assume that all stations are synchronized in time, so all chip sequences begin at the same instant. When two or more stations transmit simultaneously, their bipolar signals add linearly.e.g if in one chip period three stations output +1 and one station outputs -1, the result is +2. One can think of this as adding voltages: three stations outputting +1 volts and 1 station outputting -1 volts gives 2 volts.in the above fig2-45, we see six examples of one or more stations transmitting at the same time.In the first example, C transmits a 1 bit, so we just get C's chip sequence.In the second example, both B and C transmit 1 bit,so we get the sum of their bipolar chip sequences, namely:

(-1-1+1-1+1+1+1+1-1)+(-1+1-1+1+1+1-1-1)=(-2 0 0 0+2+2 0-2)

In the third example, station A sends a 1 and station B sends a 0. The others are silent .in the fourth example,A and C send a 1 bit while B sends a 0 bit . in the fifth example, all four stations send a 1 bit.Finally in the last example ,A,B and D send a 1 bit, while C sends a 0 bit. Each of the six sequences S1 through S6 given in Fig 2-45 (c) represents only 1 bit time.

To recover the bit stream of an individual station, the receiver must know that station's chip sequence in advance. It does the recovery by computing the normalized inner product of the received chip sequence and the chip sequence of the station whose bit stream it is trying to recover.If the received chip sequence is **S** and the receiver is trying to listen to a station whose chip sequence is **C,** it just computes the normalized inner product, **S.C.** consider two stations,A and C, both transmit a 1 bit at the same time that B transmits a 0 bit. The receiver sees the sum**, S= A +B NEG +C** and computes

S . C= (A+B NEG +C) . C= A . C+B NEG . C+ C . C = 0 + 0 + 1 =1

The first two vanish because all pairs of chip sequences have been carefully chosen to be orthogonal. That is why this property must be imposed on the chip sequences.

An alternative way of thinking about this situation is to imagine that the three chip sequences all came in separately, rather than summed. Then the receiver would compute the inner product with each one separately and add the results. Due to the orthogonality property ,all inner products except **C.C** would be 0. Adding them and then doing the inner product is in fact the same as doing the inner products and then adding those.

To make the decoding process more concrete, let us consider the six examples of fig. 2-45(c)and (d) Suppose that the receiver is interested in extracting the bit sent by station C from each of the six sums S1 through S6. It calculates the bit by summing the pairwise products of the received S and C vector of fig2-45(b) and then taking 1/8 of the result(since m=8 here).As shown the correct bit is decoded each time.

THIRD GENERATION MOBILE PHONES

Third generation systems will use the Broadband Integrated Services Digital Network(B-ISDN) to provide access to information networks, such as the internet and other public and private databases.these networks will carry many types of information(voice,data and video), will operate in varied regions(dense or sparsely populated regions),and will serve both stationary users and vehicular users travelling at high speeds. Packets radio communications will likely be used to distribute network control while providing a reliable information transfer.

The terms 3G Personal communication System(PCS)and 3G Personal communication network(PCN) are used to imply emerging third generation wireless systems for hand-held devices.Other names for PCS include Future Public Land Mobile Telecommunication Systems(FPLMTS) for worldwide use which has more recently been called International Mobile Telecommunication(IMT-2000), and Universal Mobile Telecommunication System(UMTS) for advanced mobile personal services in Europe.

Concepts of signal propagation:

Reflection , Diffraction and Scattering are the three basic propagation mechanisms which impact signal propagation.

REFLECTION occurs when a propagating electromagnetic wave impinges upon an object which has very large dimentions when compared to the wavelength if the propagating wave. Reflections occur from the surface of the earth and from buildings and walls.

DIFFRACTION occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities(edges). The secondary waves resulting from the obstructing surface are present throughout the space and even behind the obstable, giving rise to a bending of waves around the obstacle, even when a line –of- sight path does not exist between transmitter and receiver. At high frequencies,diffraction,like reflection,depends on the geometry of the object, as well as the amplitude, phase and polarization of the incident wave at the point of diffraction.

SCATTERING occurs when the medium through which the wave travels consists of objects with dimentions that are small compared to the wavelength, and where the number of obstacles per unit volume is large.Scattered waves are produced by rough surfaces, small objects,or by other regularities in the channel.In practice, foliage, street signs, and lamp posts induce scattering in a mobile communication systems.

MULTIPATH PROPAGATION : fading is used to describe the rapid fluctuations of the amplitude , phases, or multipath delays of a radio signal over a short period of time or travel distance,so that large scale path loss effects may be ignored. Fading is caused by interference between two or more versions of the transmitted signal which arrive at the receiver at slightly different times.These waves called multipath waves, combine at the receiver antenna to give a resultant signal which can vary widely in amplitude and phase, depending on the distribution of the intensity and relative propagation time of the waves and the bandwidth of the transmitted signal.the presence of reflecting objects and scatterers in the channel creates a constantly changing envoirnment that dissipates the signal energy in amplitude, phase, and time.These effects result in multiple versions of the transmitted signal that arrive at the receiving antenna, displaced with respect to one another in time and spatial orientation. The random phase and amplitudes of the different multipath components cause fluctuations in signal strength, thereby including small scale fading, signal distortion or both.Multipath propogation often lengthens the time required for the base band portion of the signalto reach the receiver which can cause signal smearing due to intersymbol interference.