# Multiple Access Techniques for Wireless Communications

Multiple access schemes are used to allow many mobile users to share simultaneously a finite amount of radio spectrum. The sharing of spectrum is required to achieve high capacity by simultaneously allocating the available bandwidth (or the available amount of channels) to multiple users. For high quality communications, this must be done without severe degradation in the performance of the system.

## Introduction

In wireless communications systems, it is often desirable to allow the subscriber to send simultaneously information to the base station while receiving information from the base station. For example, in conventional telephone systems, it is possible to talk and listen simultaneously, and this effect, called *duplexing,* is generally required in wireless telephone systems.

Duplexing may be done using frequency or time domain techniques. *Frequency division duplexing* (FDD) provides two distinct bands of frequencies for every user. The *forward band* provides traffic from the base station to the mobile, and the *reverse band* provides traffic from the mobile to the base station. In FDD, any *duplex channel*actually consists of two simplex channels (a forward and reverse), and a device called a *duplexer* is used inside each subscriber unit and base station to allow simultaneous bidirectional radio transmission and reception for both the subscriber unit and the base station on the duplex channel pair. The frequency separation between each forward and reverse channel is constant throughout the system, regardless of the particular channel being used.

*Time division duplexing* (TDD) uses time instead of frequency to provide both a forward and reverse link. In TDD, multiple users share a single radio channel by taking turns in the time domain. Individual users are allowed to access the channel in assigned *time slots,* and each duplex channel has both a forward time slot and a reverse time slot to facilitate bidirectional communication. If the time separation between the forward and reverse time slot is small, then the transmission and reception of data appears simultaneous to the users at both the subscriber unit and on the base station side. Figure 1 illustrates FDD and TDD techniques. TDD allows communication on a single channel (as opposed to requiring two separate simplex or dedicated channels) and simplifies the subscriber equipment since a duplexer is not required.

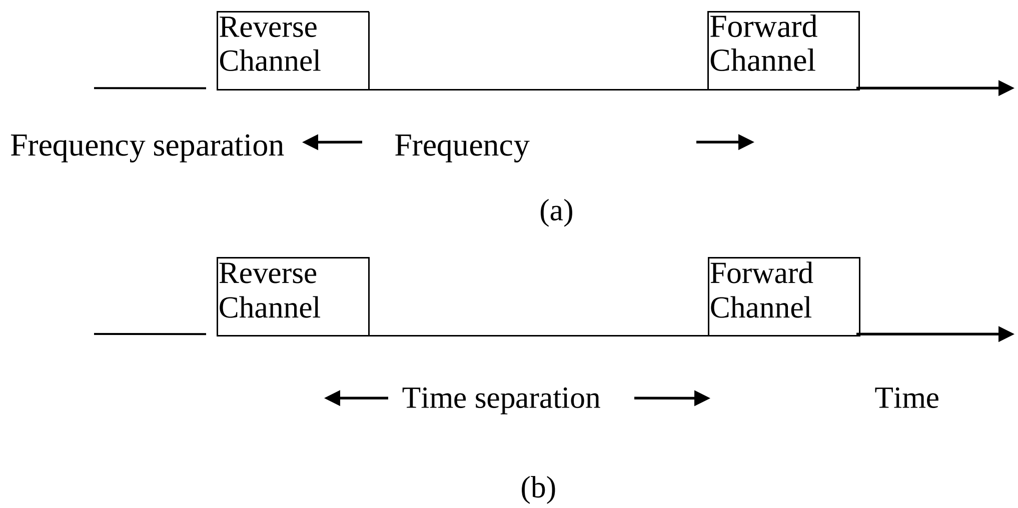



**Figure 1** (a) FDD provides two simplex channels at the same time; (b) TDD provides two simplex time slots on the same frequency.

There are several tradeoffs between FDD and TDD approaches. FDD is geared toward radio communications systems that allocate individual radio frequencies for each user. Because each transceiver simultaneously transmits and receives radio signals which can vary by more than 100 dB, the frequency allocation used for the forward and reverse channels must be carefully coordinated within its own system and with out-of-band users that occupy spectrum between these two bands. Furthermore, the frequency separation must be coordinated to permit the use of inexpensive RF and oscillator technology. TDD enables each transceiver to operate as either a transmitter or receiver on the same frequency, and eliminates the need for separate forward and reverse frequency bands. However, there is a time latency created by TDD due to the fact that communications is not full duplex in the truest sense, and this latency creates inherent sensitivities to propagation delays of individual users. Because of the rigid timing required for time slotting, TDD generally is limited to cordless phone or short range portable access. TDD is effective for fixed wireless access when all users are stationary so that propagation delays do not vary in time among the users.

# Introduction to Multiple Access

*Frequency division multiple access* (FDMA), *time division multiple access* (TDMA), and *code division multiple access* (CDMA) are the three major access techniques used to share the available bandwidth in a wireless communication system. These techniques can be grouped as *narrowband* and *wideband* systems, depending upon how the available bandwidth is allocated to the users. The duplexing technique of a multiple access system is usually described along with the particular multiple access scheme, as shown in the examples that follow.

**Narrowband Systems** — The term *narrowband* is used to relate the bandwidth of a single channel to the expected coherence bandwidth of the channel. In a narrowband multiple access system, the available radio spectrum is divided into a large number of narrowband channels. The channels are usually operated using FDD. To minimize interference between forward and reverse links on each channel, the frequency separation is made as great as possible within the frequency spectrum, while still allowing inexpensive duplexers and a common transceiver antenna to be used in each subscriber unit. In narrowband FDMA, a user is assigned a particular channel which is not shared by other users in the vicinity, and if FDD is used (that is, each duplex channel has a forward and reverse simplex channel), then the system is called FDMA/FDD. Narrowband TDMA, on the other hand, allows users to share the same radio channel but allocates a unique time slot to each user in a cyclical fashion on the channel, thus separating a small number of users in time on a single channel. For narrowband TDMA systems, there generally are a large number of radio channels allocated using either FDD or TDD, and each channel is shared using TDMA. Such systems are called TDMA/FDD or 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 does not greatly vary the received signal power within a wideband channel, and frequency selective fades occur in only a small fraction of the signal bandwidth at any instance of time. In wideband multiple access systems a large number of transmitters are allowed to transmit on the same channel. TDMA allocates time slots to the many transmitters on the same channel and allows only one transmitter to access the channel at any instant of time, whereas spread spectrum

CDMA allows all of the transmitters to access the channel at the same time. TDMA and CDMA systems may use either FDD or TDD multiplexing techniques.

In addition to FDMA, TDMA, and CDMA, two other multiple access schemes will be used for wireless communications. These are *packet radio* (PR) and *space division multiple access* (SDMA).

## Frequency Division Multiple Access (FDMA)

Frequency division multiple access (FDMA) assigns individual channels to individual users. It can be seen from Figure 2 that each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service. During the period of the call, no other user can share the same channel. In FDD systems, the users are assigned a channel as a pair of frequencies; one frequency is used for the forward channel, while the other frequency is used for the reverse channel. 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 cannot be used by other users to increase or share capacity. It is essentially a wasted resource.

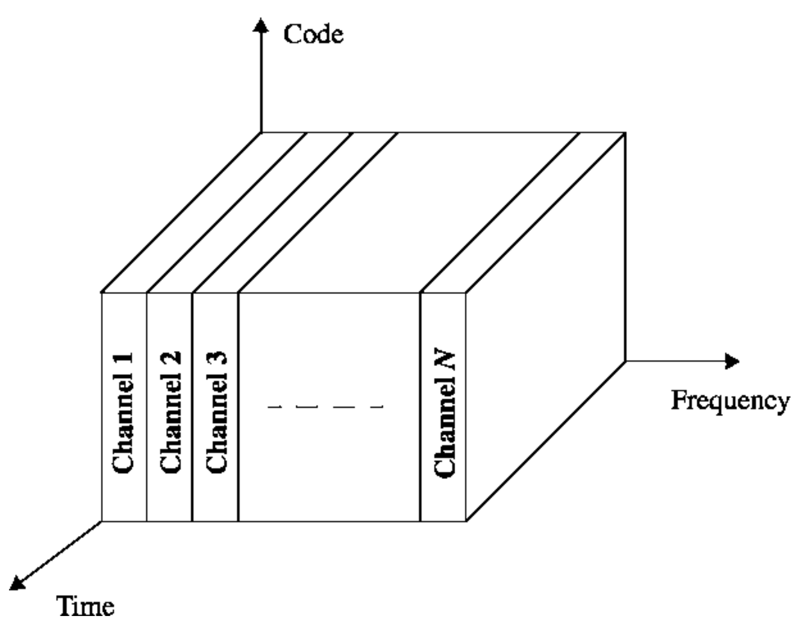



**Figure 2** FDMA where different channels are assigned different frequency bands

- After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously.
- The bandwidths of FDMA channels are relatively narrow (30 kHz in AMPS) as each channel supports only one circuit per carrier. That is, FDMA is usually implemented in narrowband systems.
- The symbol time of a narrowband signal is large as compared to the average delay spread. This implies that the amount of intersymbol interference is low and, thus, little or no equalization is required in FDMA narrowband systems.
- The complexity of FDMA mobile systems is lower when compared to TDMA systems, though this is changing as digital signal processing methods improve for TDMA.
- Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.
- FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design, and the need to use costly bandpass filters to eliminate spurious radiation at the base station.
- The FDMA mobile unit uses duplexers since both the transmitter and receiver operate at the same time. This results in an increase in the cost of FDMA subscriber units and base stations.
- FDMA requires tight RF filtering to minimize adjacent channel interference

**Nonlinear Effects in FDMA** — In a FDMA system, many channels share the same antenna at the base station. The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are nonlinear. The nonlinearities cause signal spreading in the frequency domain and generate *intermodulation* (IM) frequencies. IM is undesired RF radiation which can interfere with other channels in the FDMA systems. Spreading of the spectrum results in adjacent-channel interference. Intermodulation is the generation of undesirable harmonics. Harmonics generated outside the mobile radio band cause interference to adjacent services, while those present inside the band cause interference to other users in the wireless system .

The first US analog cellular system, the *Advanced Mobile Phone System* (AMPS), 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 channel. Voice signals are sent on the forward channel from the base station to 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. The number of channels that can be simultaneously supported in a FDMA system is given by:

$$N = \frac{B_t - 2B_{guard}}{B_c}$$

where $B_t$ is the total spectrum allocation, $B_{guard}$ is the guard band allocated at the edge of the allocated spectrum band, and $B_c$ is the channel bandwidth. Note that $B_t$ and $B_c$ may be speci-fied in terms of simplex bandwidths where it is understood that there are symmetric frequency allocations for the forward band and reverse band.

# Time Division Multiple Access (TDMA)

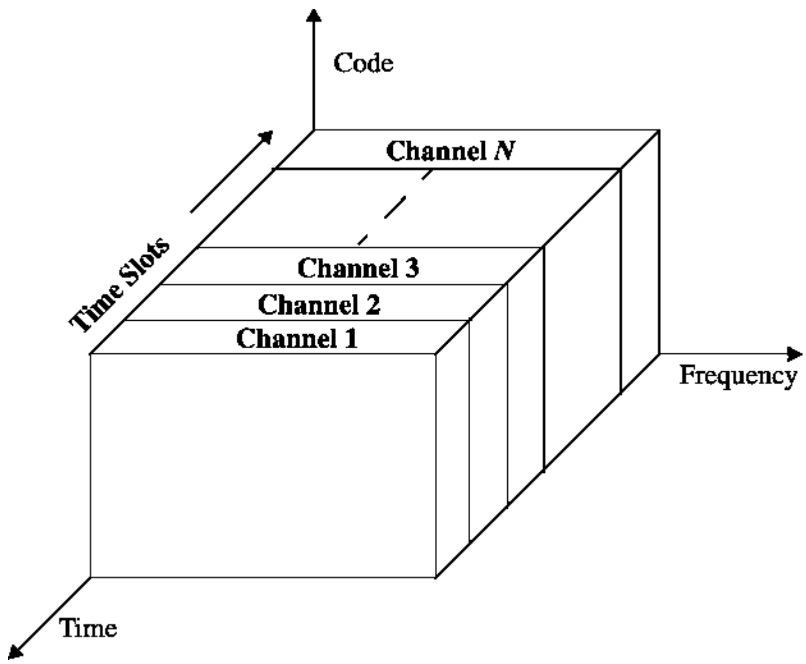



**Figure** 3 TDMA scheme where each channel occupies a cyclically repeating time slot.

*Time division multiple access* (TDMA) systems divide the radio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive. It can be seen from Figure 3 that each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot that reoccurs every frame, where *N* time slots comprise a frame. TDMA systems transmit data in a *buffer-and-burst* method, thus the transmission for any user is non-continuous. This implies that, unlike in FDMA systems which accommodate analog FM, digital data and digital modulation must be used with TDMA. The transmission from various users is interlaced into a repeating frame structure as shown in Figure 4. It can be seen that a frame consists of a number of slots. Each frame is made up of a preamble, an information message, and tail bits. In TDMA/ TDD, half of the time slots in the frame information message would be used for the forward link channels and half would be used for reverse link channels. In TDMA/FDD systems, an identical or similar frame structure would be used solely for either forward or reverse transmission, but the carrier frequencies would be different for the forward and reverse links. In general, TDMA/FDD systems intentionally induce several time slots of delay between the forward and reverse time slots for a particular user, so that duplexers are not required in the subscriber unit.

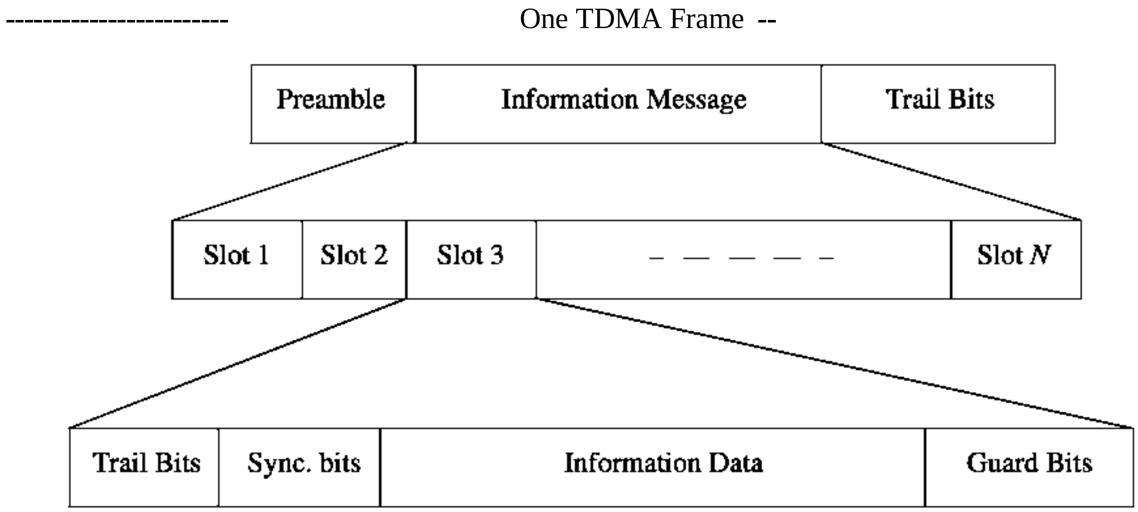



**Figure 4** TDMA frame structure. The frame is cyclically repeated over time.

In a TDMA frame, the preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other. Guard times are utilized to allow synchronization of the receivers between different slots and frames. Different TDMA standards have different TDMA frame structures.

**The features of TDMA include the following:**

- TDMA shares a single carrier frequency with several users, where each user 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 for users of a TDMA system 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 (which is most of the time).
- Because of discontinuous transmissions in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots. An enhanced link control, such as that provided by *mobile assisted handoff* (MAHO) can be carried out by a subscriber by listening on an idle slot in the TDMA frame.
- TDMA uses different time slots for transmission and reception, thus duplexers are not required. Even if FDD is used, a switch rather than a duplexer inside the subscriber unit is

all that is required to switch between transmitter and receiver using TDMA.

- Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.
- In TDMA, the guard time should be minimized. If the transmitted signal at the edges of a time slot are suppressed sharply in order to shorten the guard time, the transmitted spectrum will expand and cause interference to adjacent channels.
- High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted, and this requires the receivers to be synchronized for each data burst. In addition, guard slots are necessary to separate users, and this results in the TDMA systems having larger overheads as compared to FDMA.
- TDMA has an advantage in that it 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 slots based on priority.

***Efficiency of TDMA*** — The efficiency of a TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme. The frame efficiency $\eta_f$, is the percentage of bits per frame which contain transmitted data. Note that the transmitted data may include source and channel coding bits, so the raw end-user efficiency of a system is generally less than $\eta_f$. The frame efficiency can be found as follows.

*The number of overhead bits per frame is,*

$$\boldsymbol{b_{OH} = N_r b_r + N_t b_p + N_t b_g + N_r b_g}$$

where $N_r$ is the number of reference bursts per frame, $N_t$ is the number of traffic bursts per frame, $b_r$ is the number of overhead bits per reference burst, $b_p$ is the number of overhead bits per preamble in each slot, and $b_g$ is the number of equivalent bits in each guard time interval. The total number of bits per frame, $b_T$, is

$$b_T = T_f R$$

where $T_f$- is the frame duration, and $R$ is the channel bit rate. The frame efficiency $\eta_f$ is thus given as

$$\eta_f = \left(1 - \frac{b_{OH}}{b_T}\right) \times 100\%$$

***Number of channels in TDMA system*** — The number of TDMA channel slots that can be provided in a TDMA system is found by multiplying the number of TDMA slots per channel by the number of channels available and is given by

$$N = \frac{m(B_{tot} - 2B_{guard})}{B_c}$$

where *m* is the maximum number of TDMA users supported on each radio channel. Note that two guard bands, one at the low end of the allocated frequency band and one at the high end, are required to ensure that users at the edge of the band do not "bleed over" into an adjacent radio service.

## Spread Spectrum Multiple Access

*Spread spectrum multiple access* (SSMA) uses signals which have a transmission bandwidth that is several orders of magnitude 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 also provides immunity to multipath interference and robust multiple access capability. 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. It is exactly this situation that is of interest to wireless system designers. There are two main types of spread spectrum multiple access techniques; *frequency hopped multiple access* (FH) and *direct sequence multiple access* (DS). Direct sequence multiple access is also called *code division multiple access* (CDMA).

## Frequency Hopped Multiple Access (FHMA)

*Frequency hopped multiple access* (FHMA) is a digital multiple access system in which the carrier frequencies of the individual users are varied in a pseudorandom fashion within a wideband channel. Figure 5 illustrates how FHMA allows multiple users to simultaneously occupy the same spectrum at the same time, where each user dwells at a specific narrowband channel at a particular instance of time, based on the particular PN code of the user. The digital data of each user is broken into uniform sized bursts which are transmitted on different channels within the allocated spectrum band. The instantaneous bandwidth of any one transmission burst is much smaller than the total spread bandwidth. The pseudorandom change of the channel frequencies of the user randomizes the occupancy of a specific channel at any given time, thereby allowing for multiple access over a wide range of frequencies. In the FH receiver, a locally generated PN code is used to synchronize the receiver's instantaneous frequency with that of the transmitter. At any given point in time, a frequency hopped signal only occupies a single, relatively narrow channel since narrowband FM or FSK is used. The difference between FHMA and a traditional FDMA system is that the frequency hopped signal changes channels at rapid intervals. If the rate of change of the carrier frequency is greater than the symbol rate, then the system is referred to as *a fast frequency hopping system.* If the channel changes at a rate less than or equal to the symbol rate, it is called *slow frequency hopping.* A fast frequency hopper may thus be thought of as an FDMA system which employs frequency diversity. FHMA systems often employ energy efficient constant envelope modulation. Inexpensive receivers may be built to provide non-coherent detection of FHMA. This implies that linearity is not an issue, and the power of multiple users at the receiver does not degrade FHMA performance.

A frequency hopped system provides a level of security, especially when a large number of channels are used, since an unintended (or an intercepting) receiver that does not know the pseudorandom sequence of frequency slots must retune rapidly to search for the signal it wishes to intercept. In addition, the FH signal is somewhat immune to fading, since error control coding and interleaving can be used to protect the frequency hopped signal against deep fades which may occasionally occur during the hopping sequence. Error control coding and interleaving can also be combined to guard against *erasures* which can occur when two or more users transmit on the same channel at the same time. Bluetooth and Home RF wireless technologies have adopted FHMA for power efficiency and low cost implementation.

## Code Division Multiple Access (CDMA)

In *code division multiple access* (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the *spreading signal.* The spreading signal is a pseudo-noise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message. All users in a CDMA system, as seen from Figure 5, use the same carrier frequency and may transmit simultaneously. Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords. The receiver performs a time correlation operation to detect only the specific desired codeword. All other code words appear as noise due to de-correlation. For detection of the message signal, the receiver needs to know the codeword used by the transmitter. Each user operates independently with no knowledge of the other users.

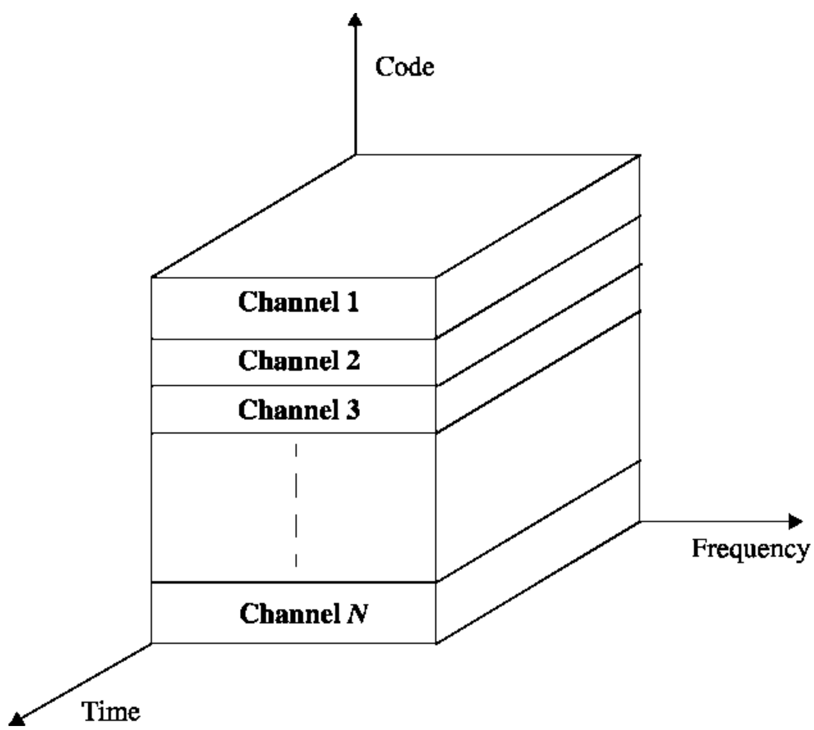



**Figure 5** Spread spectrum multiple access in which each channel is assigned a unique PN code which is orthogonal or approximately orthogonal to PN codes used by other users.

In CDMA, the power of multiple users at a receiver determines the noise floor after de-correlation. If the power of each user within a cell is not controlled such that they do not appear equal at the base station receiver, then the *near-far problem* occurs.

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will *capture* the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. To combat the near-far problem, *power control* is used in most CDMA implementations. Power control is provided by each base station in a cellular system and assures that each mobile within the base station coverage area provides the same signal level to the base station receiver. This solves the problem of a nearby subscriber overpowering the base station receiver and drowning out the signals of far away subscribers. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link. Despite the use of power control within each cell, out-of-cell mobiles provide interference which is not under the control of the receiving base station.

**The features of CDMA including the following:**

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.
- Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the number of users in a CDMA system raises the noise floor in a linear manner. Thus, there is no absolute limit on the number of users in CDMA. Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased.
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading.
- Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, multipath which is delayed by more than a chip will appear as noise. A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal.
- Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide

soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may chose the best version of the signal at any time without switching frequencies.

- Self-jamming is a problem in CDMA system. Self-jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in the de-spreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of other users in the system.
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

## Space Division Multiple Access (SDMA)

*Space division multiple access* (SDMA) controls the radiated energy for each user in space. It can be seen from Figure 8 that SDMA serves different users by using spot beam antennas. These different areas covered by the antenna beam may be served by the same frequency (in a TDMA or CDMA system) or different frequencies (in an FDMA system). Sectorized antennas may be thought of as a primitive application of SDMA. In the future, adaptive antennas will likely be used to simultaneously steer energy in the direction of many users at once and appear to be best suited for TDMA and CDMA base station architectures.

The reverse link presents the most difficulty in cellular systems for several reasons. First, the base station has complete control over the power of all the transmitted signals on the forward link. However, because of different radio propagation paths between each user and the base station, the transmitted power from each subscriber unit must be dynamically controlled to prevent any single user from driving up the interference level for all other users. Second, transmit power is limited by battery consumption at the subscriber unit, therefore there are limits on the degree to which power may be controlled on the reverse link. If the base station antenna is made to spatially filter each desired user so that more energy is detected from each subscriber, then the reverse link for each user is improved and less power is required.

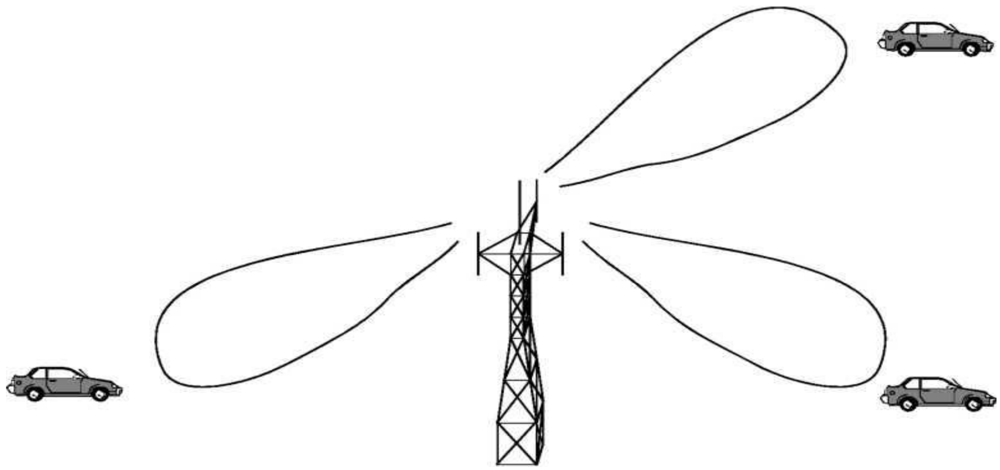


**Figure** 8 A spatially filtered base station antenna serving different users by using spot beams.

Adaptive antennas used at the base station (and eventually at the subscriber units) promise to mitigate some of the problems on the reverse link. In the limiting case of infinitesimal beam-width and infinitely fast tracking ability, adaptive antennas implement optimal SDMA, thereby providing a unique channel that is free from the interference of all other users in the cell. With SDMA, all users within the system would be able to communicate at the same time using the same channel. In addition, a perfect adaptive antenna system would be able to track individual multipath components for each user and combine them in an optimal manner to collect all of the available signal energy from each user. The perfect adaptive antenna system is not feasible since it requires infinitely large antennas.

**OFDM (Orthogonal Frequency Division Multiplexing)**

In modulations, information is mapped on to changes in frequency, phase or amplitude (or a combination of them) of a carrier signal. Multiplexing deals with allocation/accommodation of users in a given bandwidth (i.e. it deals with allocation of available resource). OFDM is a combination of modulation and multiplexing. In this technique, the given resource (bandwidth) is shared among individual modulated data sources. Normal modulation techniques (like AM, PM, FM, BPSK, QPSK, etc.., ) are single carrier modulation techniques, in which the incoming information is modulated over a single carrier. OFDM is a multicarrier modulation technique, which employs several carriers, within the allocated bandwidth, to convey the information from source to destination. Each carrier may employ one of the several available digital modulation techniques (BPSK, QPSK, QAM etc..,).

**Why OFDM**

OFDM is very effective for communication over channels with frequency selective fading ( different frequency components of the signal experience different fading). It is very difficult to handle frequency selective fading in the receiver , in which case, the design of the receiver is hugely complex. Instead of trying to mitigate frequency selective fading as a whole (which occurs when a huge bandwidth is allocated for the data transmission over a frequency selective fading channel), OFDM mitigates the problem by converting the entire frequency selective fading channel into small flat fading channels (as seen by the individual subcarriers). Flat fading is easier to combat (compared to frequency selective fading) by employing simple error correction and equalization schemes.

**Difference between FDM and OFDM:**

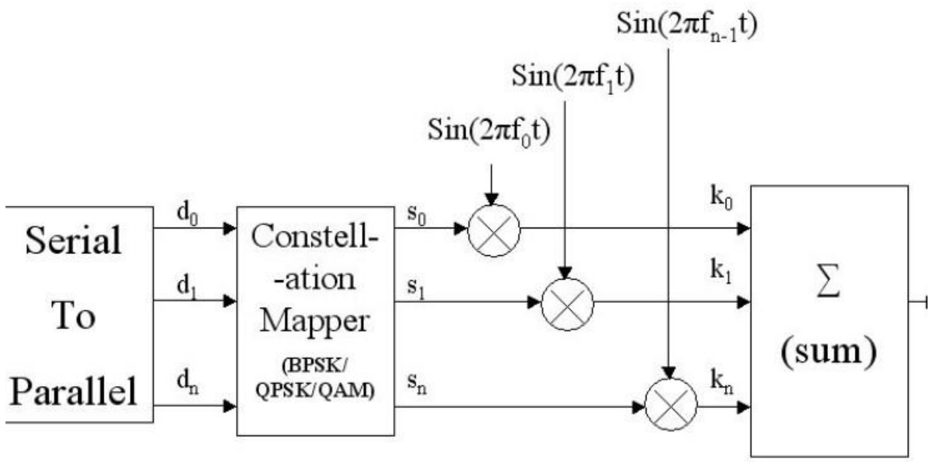
OFDM is a special case of FDM ( Frequency Division Multiplexing). In FDM, the given bandwidth is subdivided among a set of carriers. There is no relationship between the carrier frequencies in FDM. For example, consider that the given bandwidth has to be divided among 5 carriers (say a,b,c,d,e). There is no relationship between the subcarriers ;a,b,c,d and e can anything within the given bandwidth.If the carriers are harmonics, say (b=2a,c=3a,d=4a,d=5a , integral multiple of fundamental component a ) then they become orthogonal. This is a special case of FDM, which is called OFDM (as implied by the word – 'orthogonal' in OFDM)

**Designing OFDM Transmitter:**

Consider that we want to send the following data bits using OFDM : D = {d0,d1,d2,…). The first thing that should be considered in designing the OFDM transmitter is the number of subcarriers required to send the given data. As a generic case, lets assume that we have N subcarriers. Each subcarriers are centered at frequencies that are orthogonal to each other (usually multiples of frequencies).

The second design parameter could be the modulation format that we wish to use. An OFDM signal can be constructed using anyone of the following digital modulation techniques namely BPSK, QPSK, QAM etc.., .The data (D) has to be first converted from serial stream into parallel stream depending on the number of sub-carriers (N). Since we assumed that there are N subcarriers allowed for the OFDM transmission, we name the subcarriers from 0 to N-1. Now, the Serial to Parallel converter takes the serial stream of input bits and outputs N parallel streams (indexed from 0 to N-1). These parallel streams are individually converted into the required digital modulation format (BPSK, QPSK, QAM etc..,). Lets call this output $S_0$,$S_1$,..$S_N$. The conversion of parallel data (D) into the digitally modulated data (S) is usually achieved by a constellation mapper, which is essentially a look up table (LUT). Once the data bits are converted to required modulation format, they need to be superimposed on the required orthogonal subcarriers for transmission. This is achieved by a series of N parallel sinusoidal oscillators tuned to N orthogonal frequencies ($f_0, f_1, \ldots f_{N-1}$). Finally, the resultant output from the N parallel arms are summed up together to produce the OFDM signal.

The following figure illustrates the basic concept of OFDM transmission (note: In order to give a simple explanation to illustrate the underlying concept,the usual IFFT/FFT blocks that are used in actual OFDM system, are not used in the block diagram) .




OFDM Transmitter

## GSM System Architecture

The GSM system architecture 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 the *radio subsystem*,provides and manages radio transmission paths between the mobile stations and the Mobile Switching Center (MSC). The BSS also manages the radio interface between the mobile stations and all other subsystems of GSM. Each BSS consists of many Base Station Controllers (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.This subsystem interacts with the other GSM subsystems, and is provided solely for the staff of the GSM operating company which provides service facilities for the network.

Figure 1 shows the block diagram of the GSM system architecture. The Mobile Stations (MSs) communicate with the Base Station Subsystem (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.

As shown in Figure 2, 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 providers to use the same manufacturer for the BTS and BSC equipment.

The BSCs are physically connected via dedicated/leased 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. The A interface allows a service provider to use base stations and switching equipment made by different manufacturers.

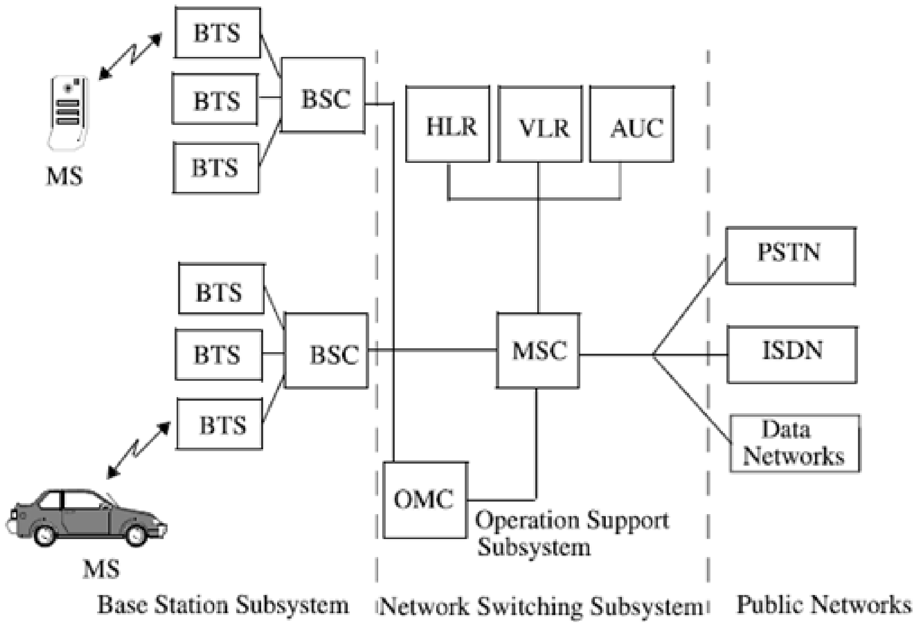



**Figure 1** GSM system architecture.

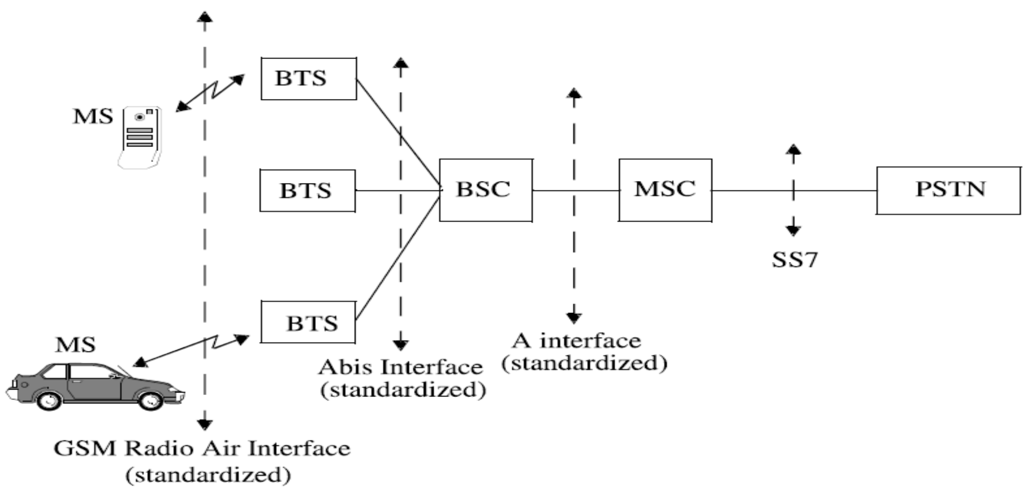



**Figure 2** The various interfaces used in GSM.

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 is 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 Location Register*(VLR), and the *Authentication Center*(AUC). The HLR is a database which contains subscriber information and location information for each user who resides in the same city as the 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 fraud-ulently 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, and 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.

**Enhanced Data Rate for GlobalEvolution (EDGE) refer the other notes**

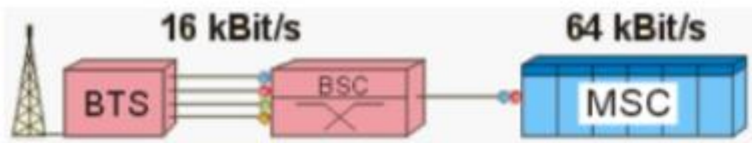
**Ultra wideband systems (UWB)**

**Push To Talk (PTT) technology refer ppt**

**Mobile IP**

**HSCSD NETWORK ARCHITECTURE**

HSCSD is High Speed Circuit Switched Data; it is 2.5G GSM standard. It is a circuit switched technique that allows a single mobile subscriber to use consecutive user time slots in GSM standard, to obtain higher speed data access on the GSM network. It relaxes the error control coding algorithms originally specified in GSM standard for data transmissions.




HSCSD is ideal for dedicated streaming internet access or real-time interactive web sessions. Increases available application data rate to 14.4 kbps. HSCSD is able to provide a raw transmission rate of up to 57.6 kbps to individual users by using up to four consecutive time slots. HSCSD bundles up to 8 GSM traffic channels into one high speed channel. HSCSD is a circuit switching technology, i.e. very suitable for constantly high data rates (e.g. telefax), but not for varying data rates (e.g. Internet browsing).

HSCSD is a feature enabling the co-allocation of multiple full rate traffic channels (TCH/F) into a HSCSD configuration. The aim of HSCSD is to provide a mixture of services with different air interface user rates by a single physical layer structure. The available capacity of a HSCSD configuration is several times the capacity of a TCH/F, leading to a significant enhancement in the air interface data transfer rate.

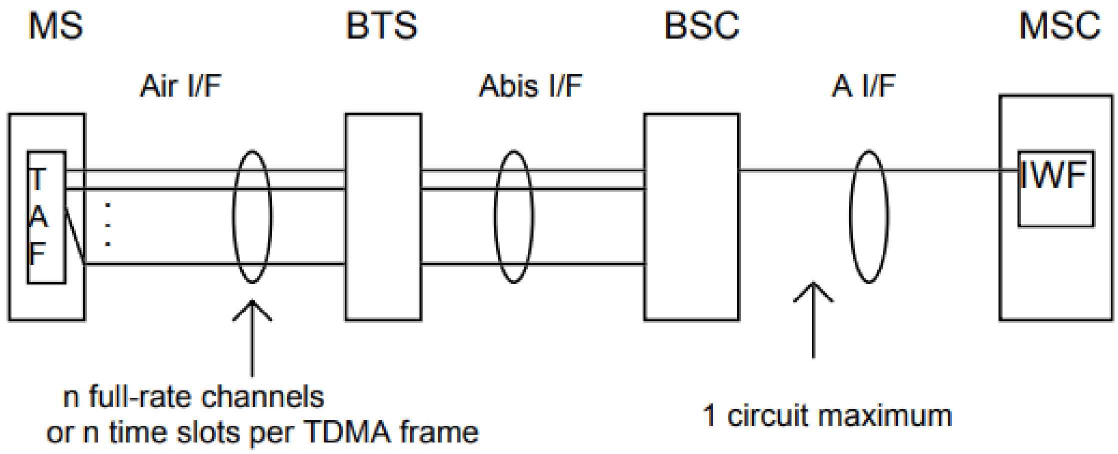



**Figure 1: Network architecture for supporting HSCSD**

A new functionality is introduced at the network and MS to provide the functions of combining and splitting the data into separate data streams which will then be transferred via n channels at the radio interface, where n = 1, 2, 3, ... 8. Once split, the data streams shall be carried by the n full rate traffic channels, called HSCSD channels, as if they were independent of each other, for the purpose of data relay and radio interface L1 error control, until to the point in the network where they are combined. However, logically the n full rate traffic channels at the radio interface belong to the same HSCSD configuration, and therefore they shall be controlled as one radio link by the network for the purpose of cellular operations, e.g. handover. This requires a new functionality in BSS.

## Digital European Cordless Telephone (DECT)

The Digital European Cordless Telephone (DECT) is a universal cordless telephone standard developed by the European Telecommunications Standards Institute. It is the first pan-European standard for cordless telephones and was finalized in July 1992.

### Features and Characteristics

- DECT provides a cordless communications framework for high traffic density, short range telecommunications, and covers a broad range of applications and environments.
- DECT offers excellent quality and services for voice and data applications .
- The main function of DECT is to provide local mobility to portable users in an in-building Private Branch Exchange (PBX). The DECT standard supports telepoint services, as well.

- DECT is configured around an open standard (OSI) which makes it possible to interconnect wide area fixed or mobile networks, such as ISDN or GSM, to a portable subscriber population.
- DECT provides low power radio access between portable parts and fixed base stations at ranges of up to a few hundredmeters.

## DECT Architecture

The DECT system is based on OSI (Open System Interconnection) principles in a manner similarto ISDN. A control plane (C-plane) and a user plane (U-plane) use the services provided bythe lower layers (i.e., the physical layer and the medium access control (MAC) layer). DECT isable to page up to 6000 subscribers without the need to know in which cell they reside (no registration required), and DECT is not a total system concept. It is designed for radio local loop or metropolitan area access, but may be used in conjunction with wide area wireless systems such as GSM. DECT uses dynamicchannel allocation based on signals received by the portable user and is specifically designed toonly support handoffs at pedestrian speeds.

**Physical Layer-**DECT uses a FDMA / TDMA/ TDD radio transmission method. Within aTDMA time slot, a dynamic selection of one out of ten carrier frequencies is used. The physicallayer specification requires that the channels have a bandwidth which is 1.5 times the channeldata rate of 1152 kbps, resulting in a channel bandwidth of 1.728 MHz. DECT has twenty-fourtime slots per frame, and twelve slots are used for communications from the fixed part to theportable (base to handset) and twelve time slots for portable to fixed (handset to base) communications.These twenty-four time slots make up a DECT frame which has a 10 ms duration. In eachtime slot, 480 bits are allocated for 32 synchronization bits, 388 data bits, and 60 bits of guardtime. The DECT TDMA time slot and frame structures are shown in Figure 11.19.

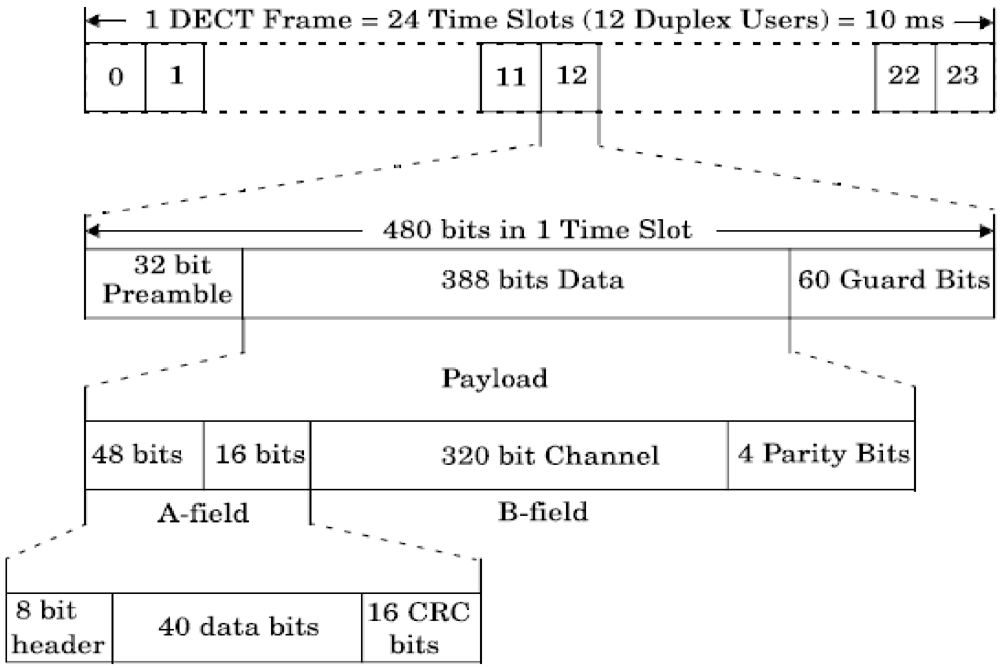



**Figure 11.19** DECT TDMA frame structure.

**Medium Access Control (MAC) Layer** - The MAC layer consists of a paging channeland a control channel for the transfer of signaling information to the C-plane. The U -plane isserved with channels for the transfer of user information (for ISDN services and frame-relay orframe-switching services). The normal bit rate of the user information channel is 32 kbps. DECT,however, also supports other bit rates. For example, 64 kbps and other multiples of 32 kbpsfor ISDN and LAN-type applications. The MAC layer also supports handover of calls and abroadcast "beacon" service that enables all idle portable units to find the best fixed radio port to
lock onto.

**Data Link Control (DLC) Layer** - The DLC layer is responsible for providing reliabledata links to the network layer and divides up the logical and physical channels into time slotsfor each user. The DLC provides formatting and error protection/correction for each time slot.

**Network Layer** - The network layer is the main signaling layer of DECT and is basedon ISDN (layer 3) and GSM protocols. The DECT network layer provides call control and circuit-switched services selected from one of the DLC services, as well as connection-orientedmessage services and mobility management.

**DECT Functional Concept**

The DECT subsystem is a microcellular or picocellular cordless telephone system that may beintegrated with or connected to a Private Automatic Branch Exchange (PABX) or to the PublicSwitched Telephone Network (PSTN). A DECT system always consists of the following fivefunctional entities as shown in Figure 11.20:

• **Portable Handset (PH)** - This is the mobile handset or the terminal. In addition, cordlessterminal adapters ( CTAs) may be used to provide fax or video communications.

• **Radio Fixed Part (RFP)** - This supports the physical layer of the DECT common airinterface. Every RFP covers one cell in the microcellular system. The radio transmissionbetween RFP and the portable unit uses multi carrier TDMA. A full duplex operation isachieved using time division duplexing (TDD).

• **Cordless Controller (CC or Cluster Controller)**- This handles the MAC, DLC, andnetwork layers for one or a cluster of RFPs and thus forms the central control unit for theDECT equipment. Speech coding is done in the CC using 32 kbps ADPCM.

• **Network-specific Interface Unit**-This supports the call completion facility in a multihandset environment. The interface recommended by the CCITT is the G.732 based onISDN protocols.

• **Supplementary Services** - This provides centralized authentication and billing whenDECT is used to provide telepoint services, and provides mobility management whenDECT is used in multilocation PABX network.

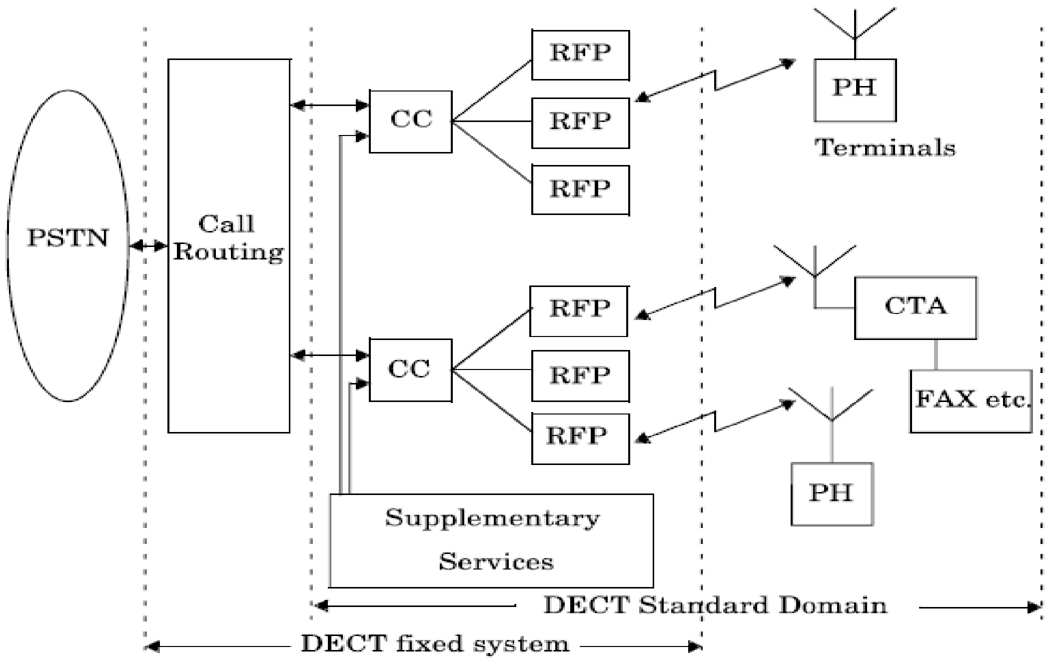



**Figure 11.20** DECT functional concept.

**DECT Radio Link**

DECT operates in the 1880 MHz to 1900 MHz band. Within this band, the DECT standarddefines ten channels from 1881.792 MHz to 1897.344 MHz with a spacing of 1728kHz. DECT supports a Multiple Carrier / TDMN / TDD structure. Each base station provides a frame structurewhich supports 12 duplex speech channels, and each time slot may occupy any of the DECTchannels. Thus, DECT base stations support FHMA on top of the TDMA / TDD structure. If the frequency hopping option is disabled for each DECT base station, a total of 120 channels withinthe DECT spectrum are provided before frequency reuse is required. Each time slot may beassigned to a different channel in order to exploit advantages provided by frequency hopping,and to avoid interference from other users in an asynchronous fashion.

**Channel Types** - DECT user data is provided in each B-field time slot (seeFigure 11.19). Three hundred twenty user bits are provided during each time slot yielding a32 kbps data stream per user. No error correction is provided although 4 parity bits are used for code error detection.

DECT control information is carried by 64 bits in every time slot of an established call (seeFigure 11.19). These bits are assigned to one of the four logical channels depending on the natureof the control information. Thus, the gross control channel data rate is 6.4 kbps per user. DECTrelies on error detection and retransmission for accurate delivery of control information. Each64 bit control word contains 16 cyclic redundancy check (CRC) bits, in addition to the 48 controldata bits. The maximum information throughput of the DECT control channel is 4.8 kbps.

**Speech Coding** - Analog speech is digitized into PCM using a 8 kHz sampling rate.The digital speech samples are ADPCM encoded at 32 kbps.

**Channel Coding** - For speech signals, no channel coding is used since DECT providesfrequency hopping for each time slot. However, thecontrol channels use 16 bit cyclic redundancy check (CRC) codes in each time slot.

**Modulation** - DECT uses a tightly filtered GMSK modulation technique. Minimum shift keying (MSK) is a form of FSK where the phase transitionsbetween two symbols are constrained to be continuous. Before the modulation, the signal isfiltered using a Gaussian shaping filter.

**Antenna Diversity** - In DECT, spatial diversity at the RFP (base station) receiver isimplemented using two antennas. The antenna which provides the best signal for each time slot
is selected. This is performed on the basis of a power measurement or alternatively by using an
appropriate quality measure (such as interference or BER). Antenna diversity helps solve fading
and interference problems. No antenna diversity is used at the subscriber unit.

## GENERAL PACKET RADIO SERVICE (GPRS)

**General Packet Radio System** is also known as **GPRS** is a third-generation step toward internet access. GPRS is also known as GSM-IP that is a Global-System Mobile Communications Internet Protocol as it keeps the users of this system online, allows to make voice calls, and access internet on-the-go. Even Time-Division Multiple Access (TDMA) users benefit from this system as it provides packet radio access. GPRS also permits the network operators to execute an Internet Protocol (IP) based core architecture for integrated voice and data applications that will continue to be used and expanded for 3G services.

GPRS supersedes the wired connections, as this system has simplified access to the packet data networks like the internet. The packet radio principle is employed by GPRS to transport user data packets in a structure way between GSM mobile stations and external packet data networks. These packets can be directly routed to the packet switched networks from the GPRS mobile stations.

In the current versions of GPRS, networks based on the Internet Protocol (IP) like the global internet or private/corporate intranets and X.25 networks are supported.

The GPRS specifications are written by the European Telecommunications Standard Institute (ETSI), the European counterpart of the American National Standard Institute (ANSI).

### Key Features

Following three key features describe wireless packet data:

- **The always online feature -** Removes the dial-up process, making applications only one click away.
- **An upgrade to existing systems -** Operators do not have to replace their equipment; rather, GPRS is added on top of the existing infrastructure.
- **An integral part of future 3G systems -** GPRS is the packet data core network for 3G systems EDGE and WCDMA.

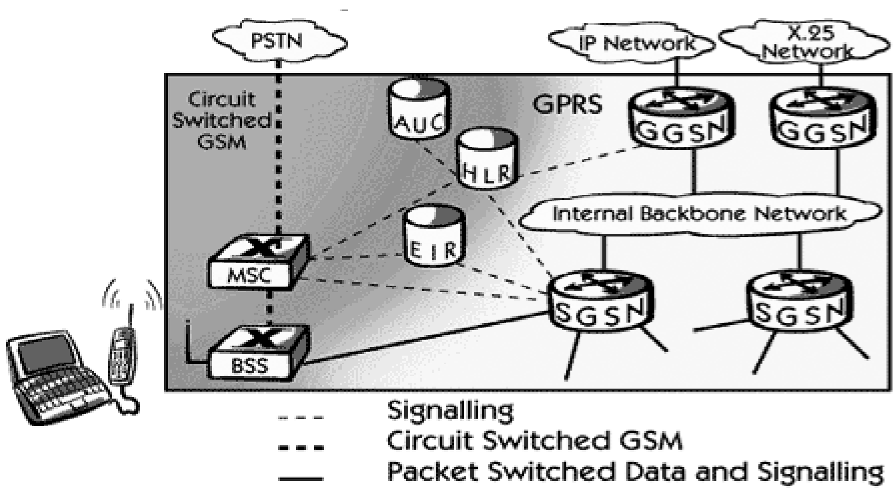
### Goals of GPRS

GPRS is the first step toward an end-to-end wireless infrastructure and has the following goals:

- Open architecture
- Consistent IP services

- Same infrastructure for different air interfaces
- Integrated telephony and Internet infrastructure
- Leverage industry investment in IP
- Service innovation independent of infrastructure

**GPRS Architecture diagram:**




GPRS attempts to reuse the existing GSM network elements as much as possible, but to effectively build a packet-based mobile cellular network, some new network elements, interfaces, and protocols for handling packet traffic are required.

Therefore, GPRS requires modifications to numerous GSM network elements as summarized below:

| **GSM Network Element** | **Modification or Upgrade Required for GPRS.** |
|---|---|
| Mobile Station (MS) | New Mobile Station is required to access GPRS services. These new terminals will be backward compatible with GSM for voice calls. |
| BTS | A software upgrade is required in the existing Base Transceiver Station(BTS). |
| BSC | The Base Station Controller (BSC) requires |

| | |
|---|---|
| | a software upgrade and the installation of new hardware called the packet control unit (PCU). The PCU directs the data traffic to the GPRS network and can be a separate hardware element associated with the BSC. |
| GPRS Support Nodes (GSNs) | The deployment of GPRS requires the installation of new core network elements called the serving GPRS support node (SGSN) and gateway GPRS support node (GGSN). |
| Databases (HLR, VLR, etc.) | All the databases involved in the network will require software upgrades to handle the new call models and functions introduced by GPRS. |

**GPRS Mobile Stations**

New Mobile Stations (MS) are required to use GPRS services because existing GSM phones do not handle the enhanced air interface or packet data. A variety of MS can exist, including a high-speed version of current phones to support high-speed data access, a new PDA device with an embedded GSM phone, and PC cards for laptop computers. These mobile stations are backward compatible for making voice calls using GSM.

**GPRS Base Station Subsystem**

Each BSC requires the installation of one or more Packet Control Units (PCUs) and a software upgrade. The PCU provides a physical and logical data interface to the Base Station Subsystem (BSS) for packet data traffic. The BTS can also require a software upgrade but typically does not require hardware enhancements.

When either voice or data traffic is originated at the subscriber mobile, it is transported over the air interface to the BTS, and from the BTS to the BSC in the same way as a standard GSM call. However, at the output of the BSC, the traffic is separated; voice is sent to the Mobile Switching Center (MSC) per standard GSM, and data is sent to a new device called the SGSN via the PCU over a Frame Relay interface.

**GPRS Support Nodes**

Following two new components, called Gateway GPRS Support Nodes (GSNs) and, Serving GPRS Support Node (SGSN) are added:

**Gateway GPRS Support Node (GGSN)**

The Gateway GPRS Support Node acts as an interface and a router to external networks. It contains routing information for GPRS mobiles, which is used to tunnel packets through the IP based internal backbone to the correct Serving GPRS Support Node. The GGSN also collects charging information connected to the use of the external data networks and can act as a packet filter for incoming traffic.

**Serving GPRS Support Node (SGSN)**

The Serving GPRS Support Node is responsible for authentication of GPRS mobiles, registration of mobiles in the network, mobility management, and collecting information on charging for the use of the air interface.

**Internal Backbone**

The internal backbone is an IP based network used to carry packets between different GSNs. Tunnelling is used between SGSNs and GGSNs, so the internal backbone does not need any information about domains outside the GPRS network. Signalling from a GSN to a MSC, HLR or EIR is done using SS7.

**Routing Area**

GPRS introduces the concept of a Routing Area. This concept is similar to Location Area in GSM, except that it generally contains fewer cells. Because routing areas are smaller than location areas, less radio resources are used While broadcasting a page message.

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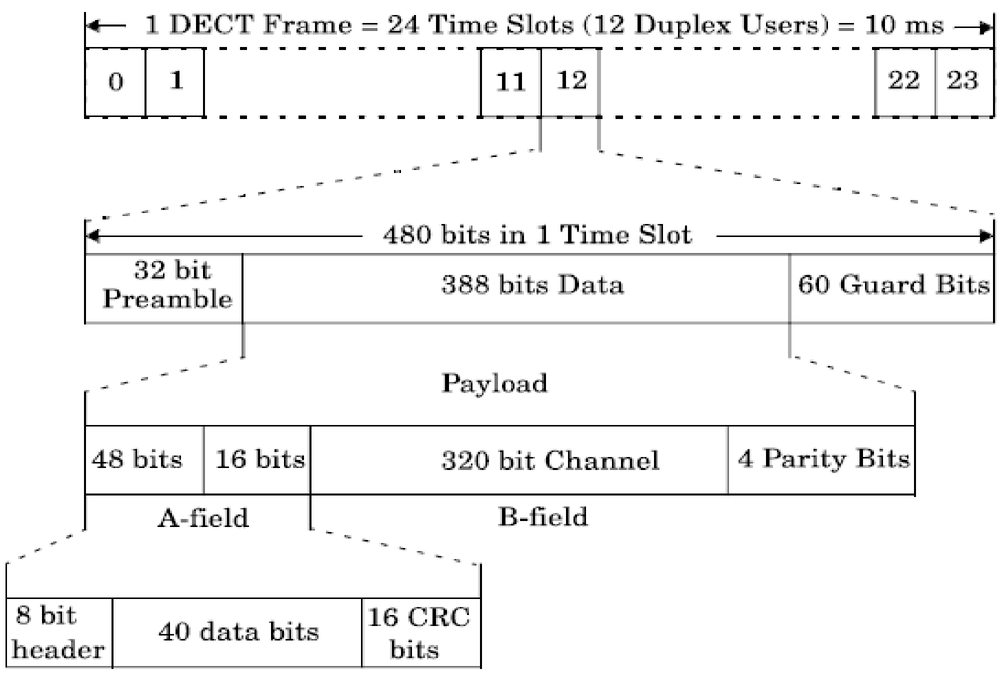


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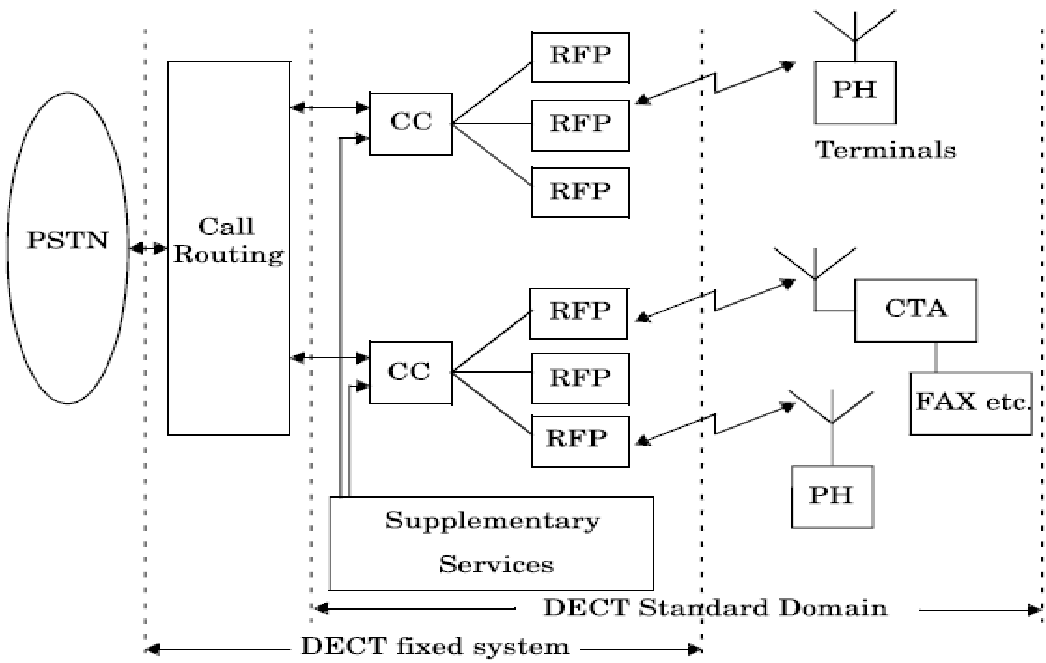


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## ENHANCED DATA RATE FOR GLOBAL EVOLUTION (EDGE)

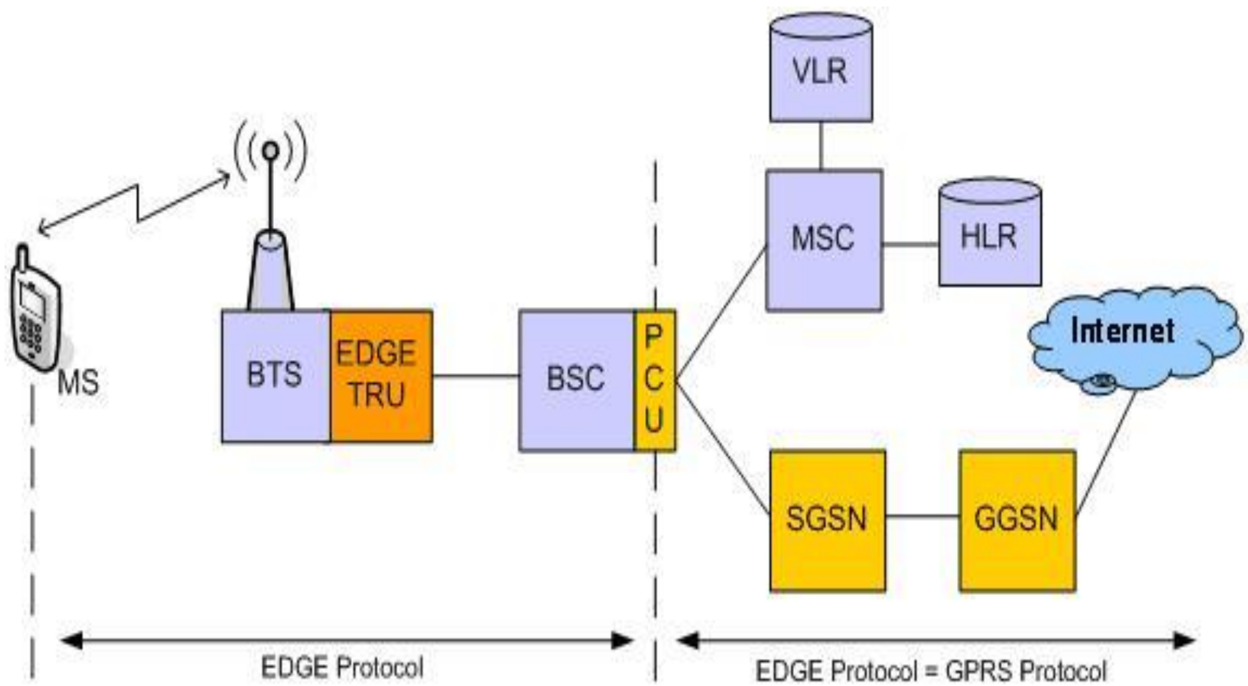
- Enhanced data for global evolution (EDGE) is a high-speed mobile data standard, it is used to enable second-generation global system for mobile communication (GSM) and time division multiple access (TDMA) networks to transmit data at up to 384 kilobits per second (Kbps).
- EDGE provides speed enhancements by changing the type of modulation used and making a better use of the carrier currently used.
- It enables a greater data-transmission speed by implementing an eight-phase-shift keying (8 PSK) modulation instead of Gaussian minimum-shift keying (GMSK).
- EDGE is a technology that gives GSM Networks the capacity to handle services for 3G.
- EDGE was developed to enable the transmission of large amounts of data at peak rates of up to 472kbps.
- Users should experience average speeds of 80 kbps to 130 kbps.
- EDGE devices are backwards compatible with GPRS.
- Although EDGE reuses the GSM carrier bandwidth and time slot structure and air interface for efficiently providing high bit rates,
- EDGE uses the same TDMA (Time Division Multiple Access) frame structure, logic channel and 200 kHz carrier bandwidth as today's GSM networks.

### Features

- It was standardized by 3GPP as a part of GSM family and was deployed in GSM networks in 2003.
- The other names for EDGE are Enhanced GPRS (EGPRS) and IMT-Single Carrier (IMT-SC).

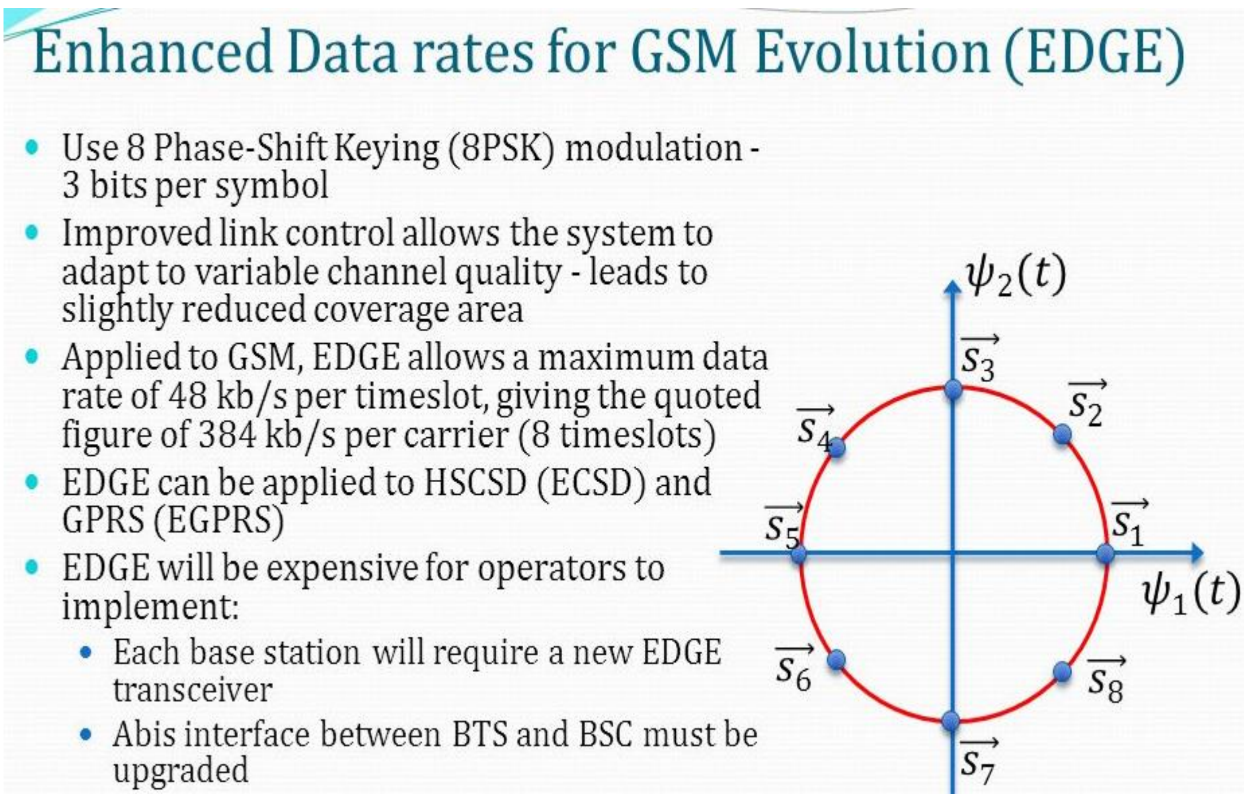
- It is compatible with any packet – switched application. It is also backward – compatible, i.e. compatible with existing or older versions.
- It enables data to be sent over a GSM TDMA systems at speeds of 384Kbps. GSM uses the modulation technique called Gaussian Minimum Shift Keying (GMSK). EDGE achieves increased bit rate by using 8PSK modulation scheme.
- In order to shift from GSM to GSM EDGE, additional network elements are incorporated to enable IP based data transfer. The two main additional nodes required are −
  - Gateway GPRS Service Node (GGSN)
  - Serving GPRS Service Node (SGSN)
- GSM EDGE compatible mobile handsets are required to avail the service.

**EDGE Architecture**




Implementation of EDGE by network operators has been designed to be simple. Only one EDGE transceiver unit will need to be added to each cell. With most vendors, it is envisaged that software upgrades to the Base Station Controller (BSCs) and Base Stations can be carried out remotely. The new EDGE capable transceiver can also handle standard GSM traffic and will automatically switch to EDGE mode when needed. Some EDGE capable terminals are expected to support high data rates in the downlink receiver only (i.e. high dates rates can be received but not sent), whilst others will access EDGE in both uplink and downlinks (i.e. high data rates can be received and sent).

The later device types will therefore need greater terminal modifications to both the receiver and the transmitter parts. EDGE is designed for migration into existing GSM and TDMA networks, enabling operators to offer multimedia and other IP-based services at speeds of up to 384 kbits/s (possibly 473 kbits/s in the future) in wide area networks. An important attraction of EDGE is the smooth evolution and upgrade of existing network hardware and software, which can be introduced into an operator's current GSM or TDMA network in existing frequency bands.



**ULTRA WIDEBAND SYSTEMS (UWB)**

As the name implies UWB, ultra wide band technology, is a form of transmission that occupies a very wide bandwidth. Typically this will be many Gigahertz, and it is this aspect that enables it to carry data rates of Gigabits per second. However the very high bandwidth used also allows the power spectral density to be very low, and the power limits on UWB are being strictly limited by the regulatory bodies.

- Ultra-Wideband (UWB) provides an interesting new technology for shortrange ultra-high speed communications in the frequency band 3.1 GHz to 10.6 GHz.
- It supports a bit rate greater than 100 Mbps within a 10-meter radius for wireless personal area communications.
- The advantages of UWB include low-power transmission, robustness for multi-path fading and low power dissipation.
- The low power transmission of the UWB is the key characteristic that might allow it to coexist with other wireless networking standards such as 802.11 LAN, 802.16 MAN and WAN.