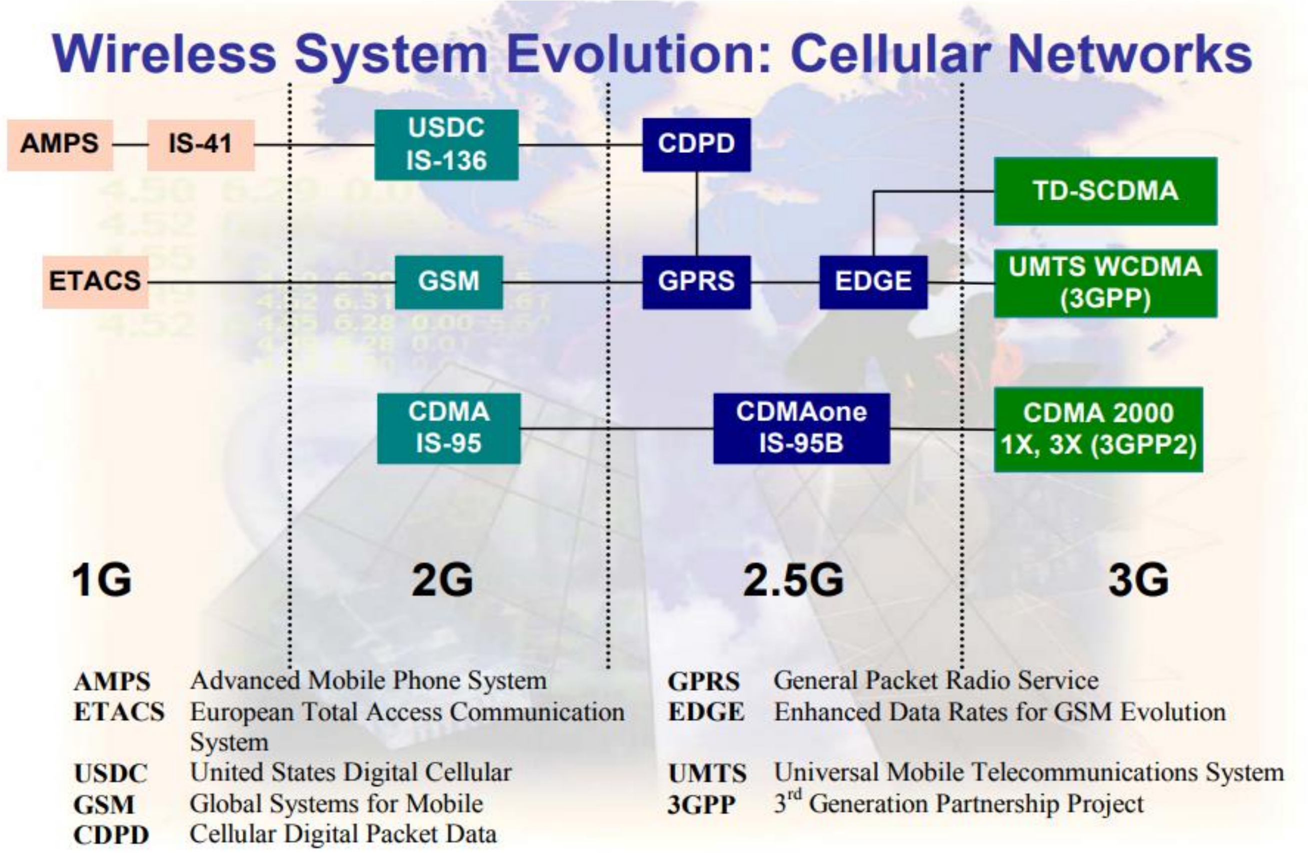
18ECC301T Wireless Communication
Unit V

Wireless systems and standards

(Notes prepared from "Wireless Communications: Principles and Practice," 2nd edition-Theodore S. Rappaport and "Wireless Communications"- Andrea Goldsmith)



Advance Mobile Phone Services (AMPS) :

In late 1970s AT &T Bell Laboratories developed the first U.S cellular telephone system called "Advanced Mobile Phone Services"

In 1983, a total of 40 MHz of spectrum in 800 MHz band was allocated by Federal Communication Commission (FCC) for AMPS

First AMPS cellular system used large cells and omni directional base station antennas to minimize initial equipment needs, and system was deployed in chicago to cover approximately 2100 square miles

AMPS system uses 7 cell reuse pattern with provision of sectoring and cell splitting to increase capacity when needed. It was found that AMPS 30KHZ channel requires a Signal to interference ration (SIR) of 18dB for satisfactory performance. The smallest reuse factor which satisfies this requirement using 120 degree is N=7 hence a 7cell reuse pattern has been adopted

Later AMPS is used throughout the world. Exact frequency allocation for AMPS differ from country to country.

European Total Access Communication (ETAC) was developed in mid 1980s and is virtually identical to AMPS except it is scaled to 25KHz and telephone number of each subscriber is formatted to accommodate country codes.

AMPS used System Identification number (SID) and ETAC used Area Identification number (AID)

AMPS and ETACS are used in 1st Generation (1G) wireless mobile communication.

First Generation-AMPS and ETAC Radio interface specification

Parameter	AMPS	ETACS
Multiple Access	FDMA	FDMA
Duplexing	FDD	FDD
Channel Bandwidth	30kHz	25kHz
Traffic Channel per RF Channel	1	1
Reverse Channel Frequency	824 – 849 MHz	890 – 915 MHz
Forward Channel Frequency	869 – 894 MHz	935 – 960 MHz
Voice Modulation	FM	FM
Peak Deviation: Voice Channels Control/Wideband Data	± 12 kHz ± 8 kHz	± 10 kHz ± 6.4 kHz
Channel Coding for Data Transmission	BCH(40,28) on FC/BCH(48,36) on RC	BCH(40,28) on FC/BCH(48,36) on RC
Data Rate on Control channel	10kbps	8kbps
Spectral Efficiency	0.33 bps/Hz	0.33 bps/Hz
Number of Channels	832	1000

AMPS Voice modulation process

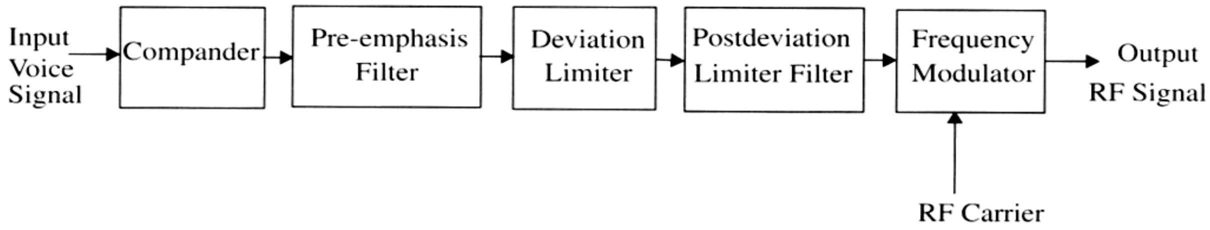


Figure 1: Block diagram of AMPS voice modulation process

Prior to frequency modulation, voice signals are processed through compander, deviation limiter and post deviation filter. Figure 1 shows the block diagram of AMPS voice modulation process. At receiver, these operations are reversed after demodulation

Compander – To accommodate large speech dynamic range, the input signals need to be compressed in amplitude range before modulation. The companding is done by 2:1 compander which produces 1 dB increase in output, for every 2dB increase in input. Companding confines the energy to 30 KHz channel bandwidth and generates a quieting effect during speech burst.

At receiver, inverse of compression is performed, thus assuring restoral of input voice level with minimum distortion

Pre-emphasis filter – Output of compressor is passed through pre-emphasis filter which increases the amplitude of high frequency bands and decrease the amplitudes of lower bands. (6 dB/octave high pass response between 0.3 KHz – 3 KHz).

Deviation Limiter – It ensures maximum frequency deviation at the mobile station is limited to ± 12 KHz . Supervisory signals and wideband data signals are excluded from this restriction.

Post Deviation Limiter Filter – The output of deviation limiter is filtered using a post deviation filer.
It is a low pass filter specified to have an attenuation of 40log10((f(in Hz)/3000)dB for frequency between 3 KHz to 5.9 KHZ and 6.1 KHZ to 15KHz
Greater than 35 dB for frequency between 5.9 to 6.1 KHz
Greater than 28 dB for frequency greater than 15KHZ

It ensures that the specifications on limitations of emission outside the specified band are met and 6 KHz Supervisory Audio Tones (SAT) which are always present during calls do not interfere with transmitted speech signal.

Global System for Mobile Communication(GSM)

GSM is a 2^{nd} generation cellular system that was developed to solve the fragmentation problems of the first cellular systems in Europe.

GSM is the world's first cellular system to specify digital modulation and network level architectures and service

The task of specifying a common mobile communication system for Europe in 900 MHz was taken up by GSM (Groupe spe'cial mobile) committee and has been renamed as Global System for Mobile Communications for marketing reasons.

GSM was introduced in European market in 1991. By end of 1993, many non-European countries had adopted GSM

Comparison of 1G and 2G standards

	1G	2G
1.	circuit switching is employed.	circuit switching as well as packet switching are employed.
2.	internet is not provided.	internet is provided.
3.	voice signal is analog signal.	Voice signal is digital signal.
4.	data services are not provided.	data services are provided except for complex data(videos).
5.	channelization protocol is FDMA.	channelization protocol is TDMA and CDMA.
6.	drawback is limited channel capacity, large phone size, low voice quality, and low battery life.	drawback is the lack of network ranges and slow data rate.
7.	AMPS(Advanced Mobile Phone System), NMT(Nordic Mobile Telephone) and TACS(Total Access Communication System) were used.	IS-95 (Interim Standard-95), GSM (Global System for Mobile communications) were used.
8.	Data rate is 2 kb/s.	Data rate is14-64 kb/s

GSM Services and Features:

GSM services are classified as

1. Teleservices
2. Data services

Teleservices includes mobile telephony and mobile originated and base originated traffic
Data services include computer to computer communication and packet switched traffic

User services are divided into 3 major categories

Telephonic Services

- Includes emergency calling and facsimile.
- GSM also supports videotex and Telex though they are not integral part of GSM

Bearer Services or Data Services

- These are limited to layer 1,2 and 3 of open system interconnection (OSI)reference model.
- Supported services include packet switched protocols and data rates from 300 bps to approx. 9.6 kbps.
- Data may be transmitted using either transparent (GSM provides standard channel coding) or non-transparent mode (GSM provides special coding based on particular data interface)

Supplementary ISDN Services

- Supplementary ISDN services, are digital in nature, and include call diversion, closed user groups, and caller identification, and are not available in analog mobile networks.
- Supplementary services also include the short messaging service (SMS) which allows GSM subscribers and base stations to transmit alphanumeric pages of limited length while simultaneously carrying normal voice traffic.
- SMS also provides cell broadcast, which allows GSM base stations to repetitively transmit ASCII messages in concatenated fashion.
- SMS may be used for safety and advisory applications, such as the broadcast of highway or weather information to all GSM subscribers within

GSM features:

- From the user's point of view, one of the most remarkable features of GSM is Subscriber Identity Module (SIM)
- SIM is a memory device that stores information such as the subscriber's identification number, the networks and countries where the subscriber is entitled to service, privacy keys, and other user-specific information.
- Without a SIM installed, all GSM mobiles are identical and non-operational.
- It is the SIM that gives GSM subscriber units their identity.

A second remarkable feature of GSM is on-the-air privacy

- Unlike analog FM cellular phone systems which can be readily monitored, it is virtually impossible to eavesdrop on a GSM radio transmission.
- The privacy is made possible by encrypting the digital bit stream sent by a GSM transmitter, according to a specific secret cryptographic key that is known only to the cellular carrier.
- This key changes with time for each user.
- Every carrier and GSM equipment manufacturer must sign the Memorandum of
- Understanding (MoU) before developing GSM equipment or deploying a GSM

GSM System Architecture:

It consists of three major interconnected subsystems that interact between themselves.

- Base Station Subsystem (BSS)
- Network and Switching Subsystem (NSS)
- Operation Support Subsystem (OSS)

Figure 2 shows the block diagram of the GSM system architecture.

Mobile Station (MS)

- It is also a subsystem, but is usually considered to be part of the BSS for architecture purposes.
- It communicate with the Base Station Subsystem (BSS) over the radio air interface

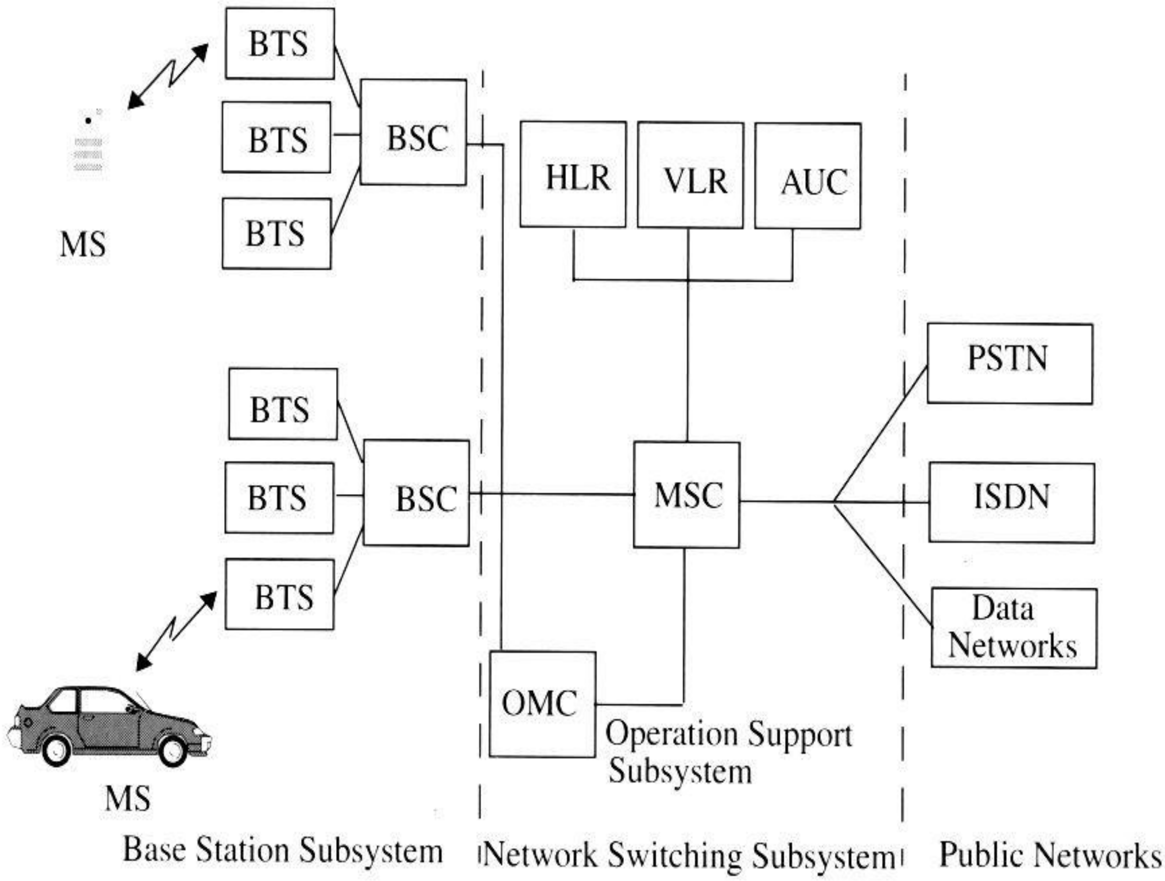


Figure 2: Block diagram of GSM architecture

Base Station Subsystem (BSS)

- It is also known as the radio subsystem, provides and manages radio transmission paths between the mobile stations and the Mobile Switching Center(MSC).
- It 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 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 maybe 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 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.

Network and Switching Subsystem (NSS)

- The NSS manages the switching functions of the system and allows the MSCs to communicate with other networks such as the PSTN and ISDN.
- Handles the switching of GSM calls between external networks and the BSCs in the radio subsystem.
- 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.
- NSS contains three different data bases:
  - Home Location Register (HLR)
  - Visitor Location Register (VLR)
  - Authentication Center (AUC)

Home Location Register:

- 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.

Visiting Location Register:

- Temporarily stores the IMSI and customer information for each roaming subscriber who is visiting the coverage area of a particular MSC.
- If 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.

Authentication Center:

- A strongly protected database which handles the authentication and encryption keys for every single subscriber in the HLR and VLR.
- AUC has Equipment Identity Register (EIR).
- EIR identifies stolen or fraudulently altered phones whose identities are not there in VLR or HLR.

Operation Support Subsystem (OSS)

- The OSS supports the operation and maintenance of GSM and allows system engineers to monitor, diagnose, and troubleshoot all aspects of the GSM system.
- Supports one or several Operation Maintenance Centers (OMC).
- OMC used to monitor and maintain the performance of each MS, BS, BSC, and MSC within a GSM system.

The OSS has three main functions:

- Maintain all telecommunications hardware and network operation.
- Manage all charging and billing procedures.
- Manage all mobile equipment in the system.

<u>GSM interface:</u>

The various types of GSM interface is shown in Figure 3.

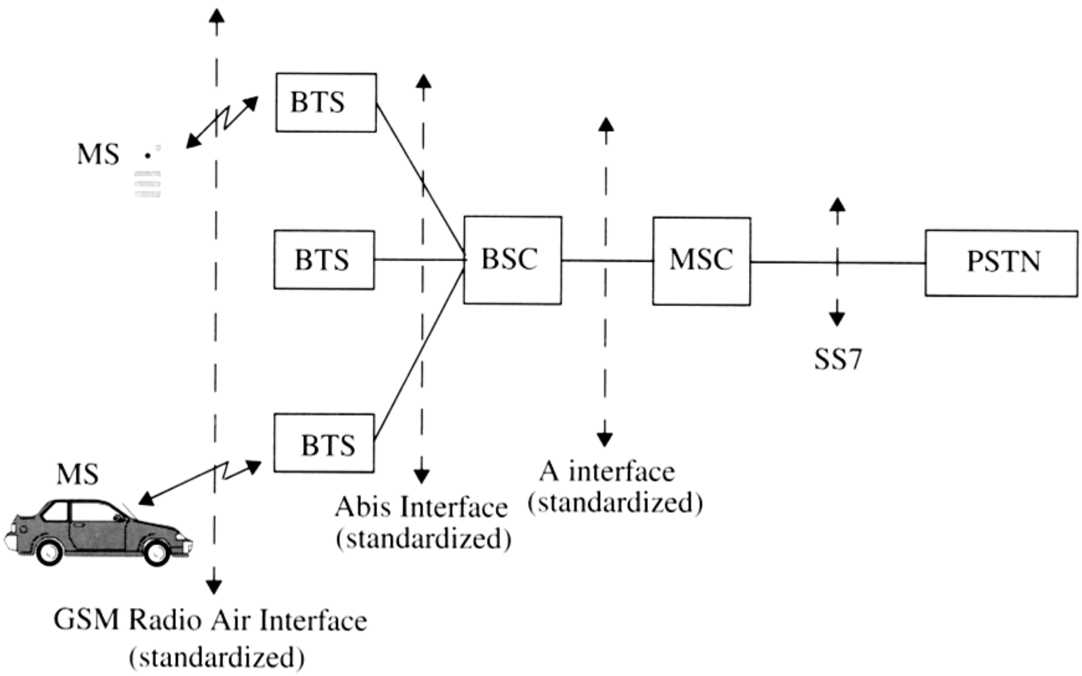


Figure 3: Various interfaces used in GSM

- 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.

GSM Radio Subsystem

- GSM utilizes two bands of 25 MHz which have been set aside for system use in all member countries.
- 890-915 MHz band is used for subscriber-to base transmissions (reverse link),
- 935-960 MHz band is used for base-to subscriber transmissions (forward link).
- The available forward and reverse frequency bands are divided into 200 kHz wide channels called ARFCNs (Absolute Radio Frequency Channel Numbers).
- The ARFCN denotes a forward and reverse channel pair which is separated in frequency by 45 MHz and each channel is time shared between as many as eight subscribers using TDMA.
- Each of the eight subscribers uses the same ARFCN and occupies a unique timeslot (TS) per frame.
- Radio transmissions on both the forward and reverse link are made at a channel data rate of 270.833 kbps) using binary BT=0.3 GMSK modulation. Thus, the signaling bit duration is 3.692 and the effective channel transmission rate per user is 33.854 kbps (270.833 kbps/8 users).
- With GSM overhead, user data is actually sent at a maximum rate of 24.7 kbps.
- Each TS has an equivalent time allocation of 156.25 channel bits as shown in Figure 4.Out of this, 8.25 bits of guard time and 6 total start and stop bits are provided to prevent overlap with adjacent time slots.
- Each TS has a time duration of 576.92 μs and a single GSM TDMA frame spans 4.615 ms. The total number of available channels within a 25 MHz bandwidth is 125 (assuming no guard band).
- Since each radio channel consists of 8 time slots, there are thus a total of 1000 traffic channels within GSM.
- In practical implementations, a guard band of 100 kHz is provided at the upper and lower end of the GSM spectrum, and only 124 channels are implemented

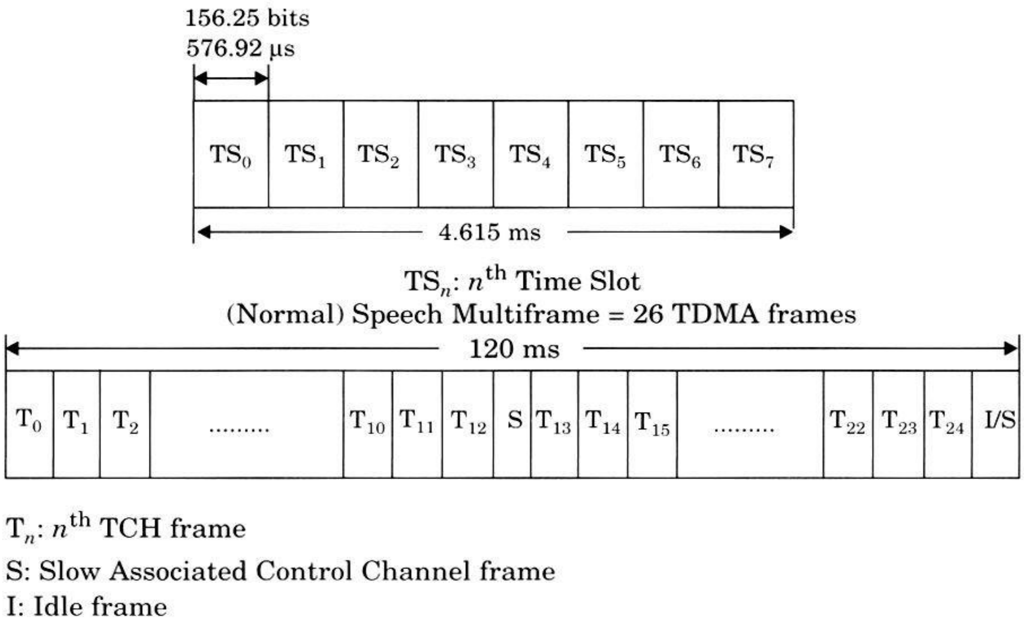


Figure 4: Speech dedicated control channel frame and multiframe structure

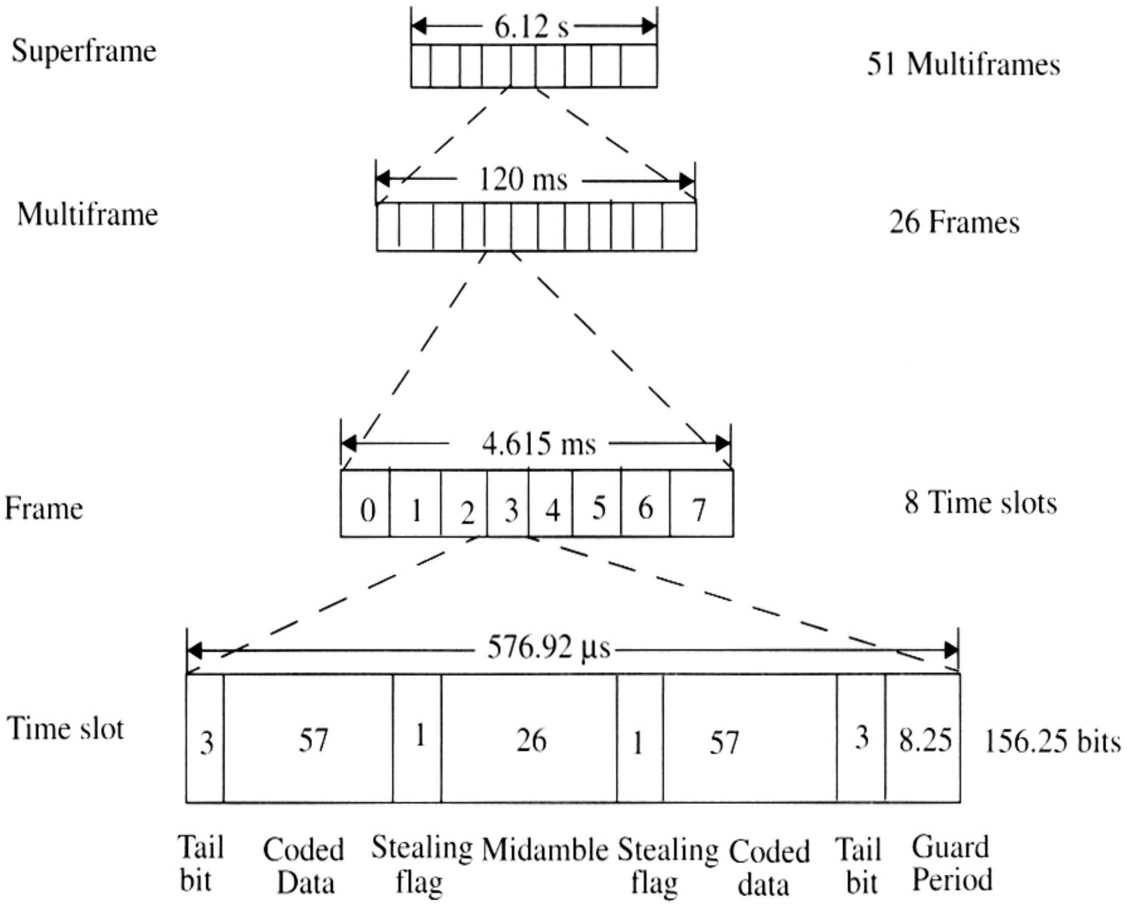


Figure 5: GSM frame structure

Figure 5 illustrates the data structure within a normal burst. It consists of 148 bits which are transmitted at a rate of 270.833333 kbps (an unused guard time of 8.25 bits is provided at the end of each burst).

- Out of the total 148 bits per TS, 114 are information-bearing bits which are transmitted as two 57 bit sequences close to the beginning and end of the burst.
- The midamble consists of a 28 bit training sequence which allows the adaptive equalizer in the mobile or base station receiver to analyze the radio channel characteristics before decoding the user data.
- On either side of the midamble there are control bits called stealing flags.
- These two flags are used to distinguish whether the TS contains voice or control data, both which share the same physical channel.
- During a frame, a GSM subscriber unit uses one TS to transmit, one TS to receive, and may use the six spare time slots to measure signal strength on five adjacent base stations as well as its own base station
- Each of the normal speech frames are grouped into larger structures called multiframes which in turn are grouped into superframes and hyperframes (hyperframes are not shown in Figure 5).
- One multiframe contains 26 TDMA frames, and one superframe contains 51 multiframes, or 1326 TDMA frames.
- A hyperframe contains 2048 superframes,

GSM Air Interface Specifications Summary

Parameter	Specifications
Reverse Channel Frequency	890–915 MHz
Forward Channel Frequency	935–960 MHz
ARFCN Number	0 to 124 and 975 to 1023
Tx/Rx Frequency Spacing Tx/Rx Time Slot Spacing	45 MHz 3 Time slots
Modulation Data Rate	270.833333 kbps
Frame Period	4.615 ms
Users per Frame (Full Rate)	8
Time Slot Period	576.9 μs
Bit Period	3.692 μs
Modulation	0.3 GMSK
ARFCN Channel Spacing	200 kHz
Interleaving (max. delay)	40 ms
Voice Coder Bit Rate	13.4 kbps

Problem: If GSM uses a frame structure where each frame consists of S time slots, and each time slot contains 156.25 bits, and data is transmitted at 270.833 kbps in the channel, find (a) the time duration of a bit, (b) the time duration of a slot,(c) the time duration of a frame, and (d) how long must a user occupying a single time slot must wait between two simultaneous transmissions.

Solution:

(a) The time duration of a bit, T_b= 1/270.833 kbps= 3.692 μs

(b) The time duration of a slot, T_{slot}= 156.25 x T_b = 0.577 ms.

(c) The time duration of a frame, T_f, = 8 x T_{slot}= 4.615 ms.

(d) A user has to wait 4.615 ms, the arrival time of a new frame, for its next transmission

Problem: If a normal GSM time slot consists of 6 trailing bits, 8.25 guard bits, 26 training bits, and 2 traffic bursts of 58 bits of data, find the frame efficiency.

Solution:

A time slot has 6 + 8.25 + 26 + 2(58) = 156.25 bits.

A frame has 8x 156.25 = 1250 bits/frame.

The number of overhead bits per frame is given by

B_{OH} = 8(6) + 8(8.25) ÷ 8(26) = 322 bits

Thus, the frame efficiency η_F = (1- (322/1250)) X100= 74.24%

GSM speech operations input and output

Figure 6 shows GSM operations from speech input to speech output

Speech Coding — The GSM speech coder is based on the Residually Excited Linear Predictive Coder (RELP), which is enhanced by including a Long-Term Predictor (LTP) .
The coder provides 260 bits for each 20 ms blocks of speech, which yields a bit rate of 13kbps

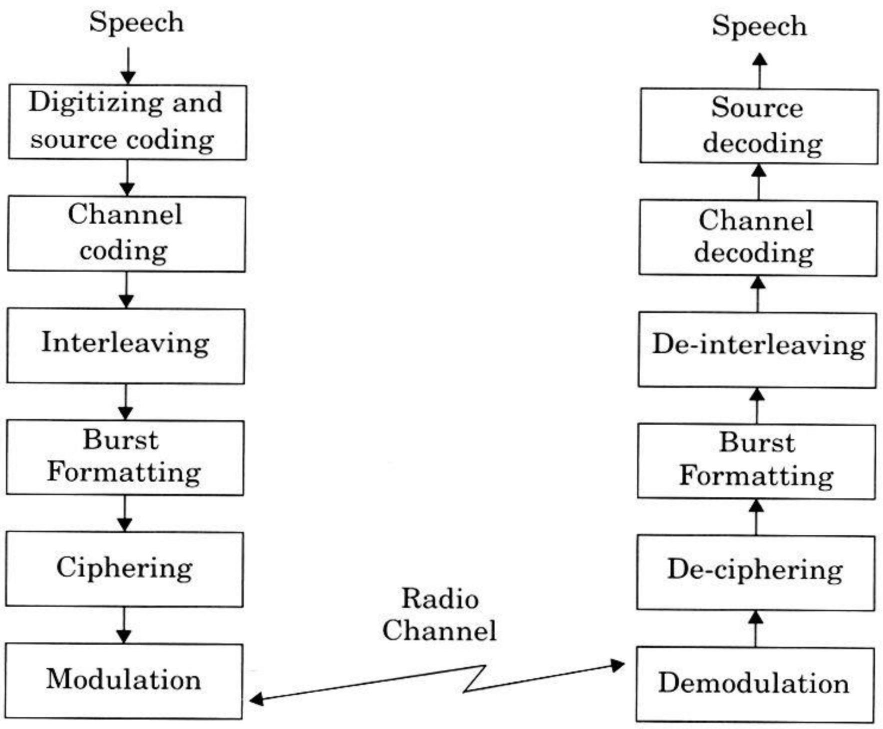


Figure 6: GSM operations from speech input to speech output

Channel Coding

- The output bits of the speech coder are ordered into groups for error protection, based upon their significance in contributing to speech quality.
- Out of the total 260 bits in a frame, the most important 50 bits, called type I a bits, have 3 parity check (CRC) bits added to them as shown in Figure 7. This facilitates the detection of non-correctable errors at the receiver.
- The next 132 bits along with the first 53 (50 type I a bits + 3 parity bits) are reordered and appended by 4 trailing zero bits, thus providing a data block of 189 bits.
- This block is then encoded for error protection using a rate 112 convolutional encoder with constraint length K = 5, thus providing a sequence of 378 bits.
- The least important 78 bits do not have any error protection and are concatenated to the existing sequence to form a block of 456 bits in a 20 ms frame.

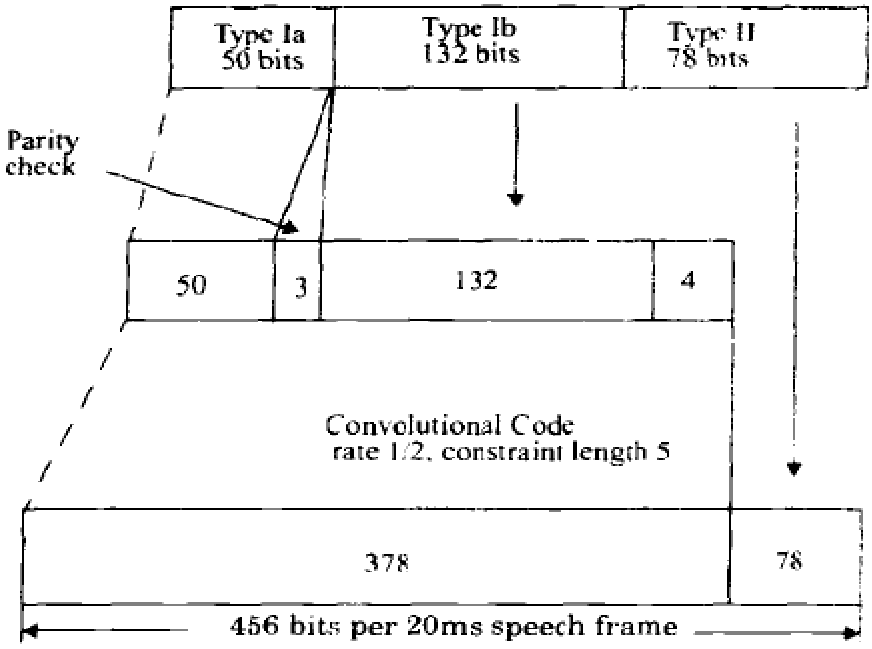


Figure 7: Error protection for signals in GSM

Channel Coding for Data Channels

- The coding provided for GSM full rate data channels is based on handling 60 bits of user data at 5ms intervals
- 240 bits of user data are applied with 4 tailing bits to a half-rate punctured convolutional coder with constraint length K = 5.
- The resulting 488 coded bits are reduced to 456 encoded data bits through puncturing (32 bits are not transmitted), and the data is separated into four 114 bit data bursts that are applied in an interleaved fashion to consecutive time slots.

Channel Coding for Control Channels

- GSM control channel messages are defined to be 184 bits long, and are encoded using a shortened binary cyclic fire code, followed by a half-rate convolutional coder.

The fire code uses the generator polynomial

$$G_5(x) = (x^{23} + 1)(x^{17} + x^3 + 1) = x^{40} + x^{26} + x^{23} + x^{17} + x^3 + 1$$

which produces 184 message bits, followed by 40 parity bits.

- Four tail bits are added to clear the convolutional coder which follows, yielding a 228 bit data block.
- This block is applied to a half-rate K = 5 convolutional code which uses the generator polynomials $G_0(x) = 1+x^3+x^4$ and $G_1(x) = 1+x+x^3+x^4$

Interleaving

- In order to minimize the effect of sudden fades on the received data, the total of 456 encoded bits within each 20 ms speech frame or control message frame are broken into eight 57 bit sub-blocks.
- These eight subblocks which make up a single speech frame are spread over eight consecutive Traffic channel (TCH) time slots.
- If a burst is lost due to interference or fading, channel coding ensures that enough bits will still be received correctly to allow the error correction to work. Each TCH time slot carries two 57 bit blocks of data from two different 20 ms speech segments.

Ciphering

- Ciphering modifies the contents of the eight interleaved blocks through the use of encryption techniques known only to the particular mobile station and base transceiver station.
- Security is further enhanced by the fact that the encryption algorithm is changed from call to call. Two types of ciphering algorithms, called A3 and AS
- The A3 algorithm is used to authenticate each mobile by verifying the users passcode within the SIM with the cryptographic key at the MSC.
- The AS algorithm provides the scrambling for the 114 coded data bits sent in each time slot.

Burst Formatting

- Burst formatting adds binary data to the ciphered blocks, in order to help synchronization and equalization of the received signal.

Modulation

- The modulation scheme used by GSM is 0.3 GMSK where 0.3 describes the 3 dB bandwidth of the Gaussian pulse shaping filter with relation to the bit rate
- GMSK is a special type of digital FM modulation.
- This minimizes the bandwidth occupied by the modulation spectrum and hence improves channel capacity. The MSK modulated signal is passed through a Gaussian filter to smooth the rapid frequency transitions which would otherwise spread energy into adjacent channels.

Frequency Hopping

- Under normal conditions, each data burst belonging to a particular physical channel is transmitted using the same carrier frequency.
- However, if users in a particular cell have severe multipath problems, the cell may be defined as a hopping cell by the network operator, in which case slow frequency hopping may be implemented to combat the multipath or interference effects in that cell.
- Frequency hopping is completely specified by the service provider.

Equalization

- Equalization is performed at the receiver with the help of the training sequences transmitted in the midamble of every time slot.
- The type of equalizer for GSM is not specified and is left up to the manufacturer.

Demodulation

- The appropriate TS is demodulated with the aid of synchronization data provided by the burst formatting.
- After demodulation, the binary information is deciphered, de-interleaved, channel decoded, and speech decoded.

Code Division Multiple Access (CDMA)

- A digital cellular system based on CDMA which promises increased capacity and has been standardized as Interim Standard 95 (IS-95) by the U.S Telecommunications Industry Association (TIA).
- IS-95 allows each user within a cell to use the same radio channel, and users in adjacent cells also use the same radio channel, since this is a direct sequence spread spectrum CDMA system.
- The IS-95 system is designed to be compatible with the existing U.S. analog cellular system (AMPS) frequency band, hence mobiles and base stations can be economically produced for dual mode operation.
- To facilitate graceful transition from AMPS to CDMA, each IS-95 channel occupies 1.25 MHz of spectrum on each one-way link.
- A pilot code is also transmitted simultaneously and at a higher power level, thereby allowing all mobiles to use coherent carrier detection while estimating the channel conditions.
- On the reverse link, all mobiles respond in an asynchronous fashion and have ideally a constant signal level due to power control applied by the base station.
- The speech coder used in the 15-95 system is the Qualcomm 9600 bps Code Excited Linear Predictive (QCELP) coder.
- The original implementation of this vocoder detects voice activity, and reduces the data rate to 1200 bps during silent periods. Intermediate user data rates of 2400, 4800, and 9600 bps.

Advantages of CDMA

1. Frequency diversity

Frequency-dependent transmission impairments (noise bursts, selective fading) have less effect

2. Multipath resistance

DSSS overcomes multipath fading by frequency diversity. Also, chipping codes used only exhibit low cross correlation and low autocorrelation

Version of signal delayed more than one chip interval does not interfere with the dominant signal as much

3. Data Privacy

Disadvantages of CDMA

1. With CDMA, as more users access the system simultaneously, noise level and hence error rate increases .Gradually system degrades
2. Self-jamming
Unless all mobile users are perfectly synchronized, arriving transmissions from multiple users will not be perfectly aligned on chip boundaries
Spreading sequences of different users not orthogonal
3. Near-far problem
Signals closer to receiver are received with less attenuation than signals farther away
Given lack of complete orthogonality, transmissions from more remote mobile units may be more difficult to recover

Frequency and Channel Specifications of Interim Standard (IS-95)

- Reverse link operation in the 824 - 849 MHz band and 869 - 894 MHz for the forward link.
- A forward and reverse channel pair is separate by 45 MHz. Many users share a common channel for transmission.
- The maximum user data rate is 9.6 kb/s. User data in IS-95 is spread to a channel chip rate of 1.2288 Mchip/s (a total spreading factor of 128) using a combination of techniques.
- The spreading process is different for the forward and reverse links.

The IS-95 channel structure is shown in Figure 8.

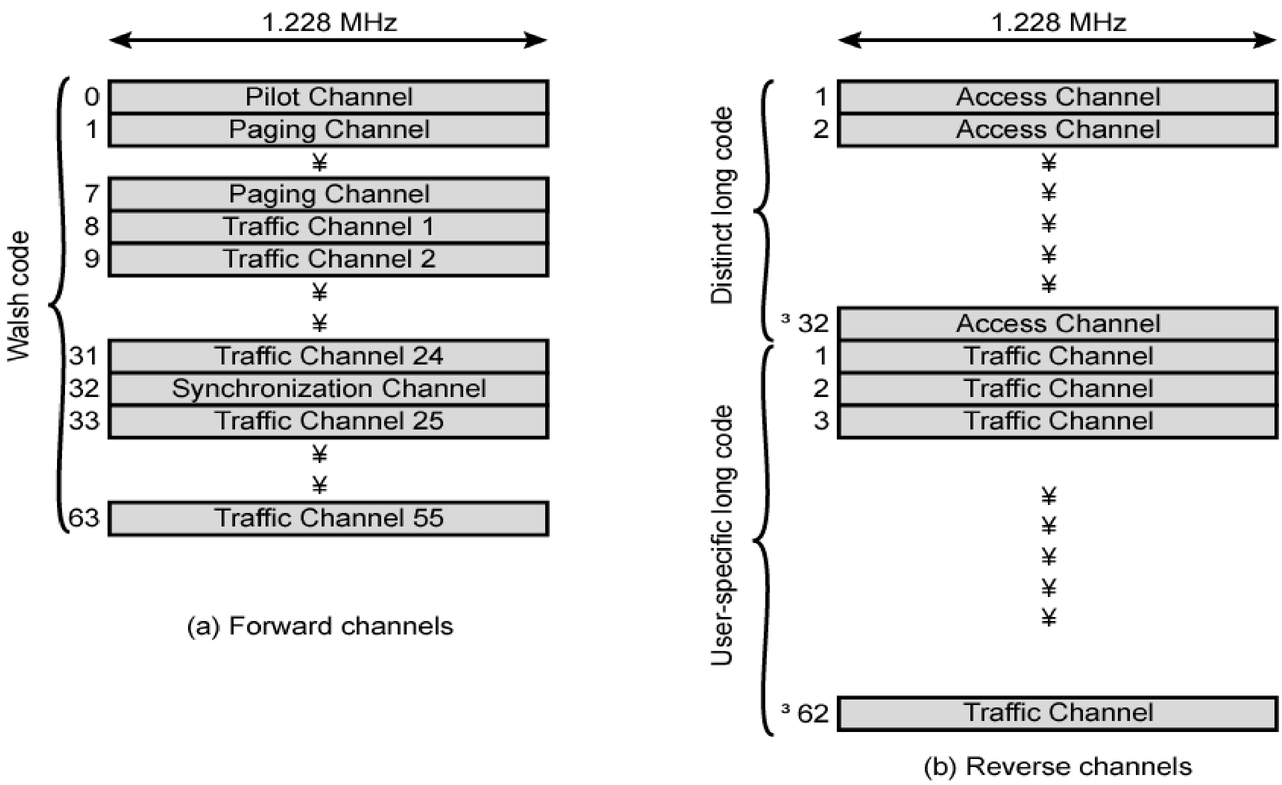


Figure 8: IS-95 channel structure

IS -95 Forward channel structure

Up to 64 logical CDMA channels each occupying the same 1228-kHz bandwidth
Four types of channels:

Pilot (channel 0)

- Continuous signal on a single channel
- allows a mobile station to acquire timing for the Forward CDMA channel
- Provides phase reference for demodulation process
- provides each mobile with a means for signal strength comparisons between base stations for determining when to handoff.
- Consists of all zeros

Synchronization (channel 32)

- 1200-bps channel used by mobile station to obtain identification information about the cellular system
- provides each mobile with a means for signal strength comparisons base stations for determining when to handoff.

Paging (channels 1 to 7)

- Contain messages for one or more mobile stations

Traffic (channels 8 to 31 and 33 to 63)

- 55 traffic channels
- Original specification supported data rates of up to 9600 bps
- Revision added rates up to 14,400 bps

IS-95 forward channel modulation system:

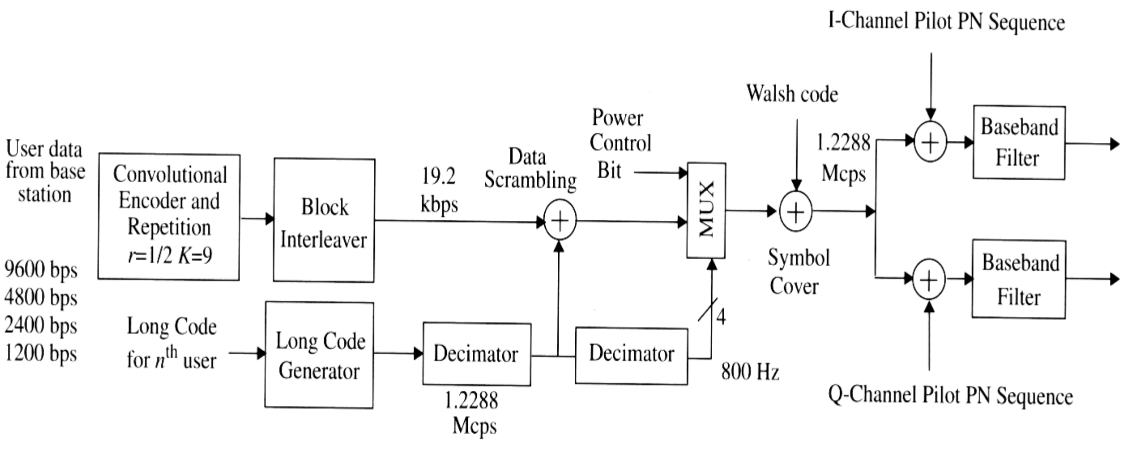


Figure 9: IS-95 Forward channel modulation system

The block diagram of IS-95 Forward channel modulation system is shown in Figure 9.

- The speech data rate applied to the transmitter is variable over the range of 1200 bps to 9600 bps.
- Speech coded voice or user data are encoded using a half-rate convolutional encoder with constraint length 9. The encoding process is described by generator vectors and G1 which are 753 (octal) and 561 (octal).
- The speech encoder exploits pauses and gaps in speech, and reduces its output from 9600 bps to 1200 bps during silent periods.
- In order to keep a constant baseband symbol rate of 19.2 kbps, whenever the user rate is less than 9600 bps, each symbol from the convolution encoder is repeated before block interleaving.

Block Interleaver

- After convolution coding and repetition, symbols are sent to a 20ms block interleaver which is a 24 by 16 array.

Data scrambling

- After Block interleaver, data is scrambled by long PN sequence code.
- The 1.2288 MHz PN sequence is applied to a decimator, which keeps only the first chip out of every sixty-four consecutive PN chips.
- The symbol rate from the decimator is 19.2 kbps.
- The data scrambling is performed by modulo-2 addition of the interleaver output with the decimator output symbol

Long PN sequence

- In the forward channel, direct sequence is used for data scrambling.
- uniquely assigned to each user is a periodic long code with period $2^{42}-1$ chips
- Each PN chip of the long code is generated by the modulo-2 inner product of a 42 bit mask and the 42 bit state vector of the sequence generator.
- Output of long code generator after decimator is at a rate of 1.2288 Mbps
- The initial state of the generator is defined to be when the output of the generator becomes '1' after following 41 consecutive '0' outputs, with the binary mask consisting of '1' in the most significant bit (MSB) followed by 41 '0's.
- Two types of masks are used in the long code generator: a public mask for the mobile station's electronic serial number (ESN) and a private mask for the mobile station identification number (MIN).
- All CDMA calls are initiated using the public mask.
- Transition to the private mask is carried out after authentication is performed.
- The long code is specified by the following characteristic polynomial is given below

$$p(x) = x^{42} + x^{35} + x^{33} + x^{31} + x^{27} + x^{26} + x^{25} + x^{22} + x^{21} + x^{19} + x^{18} + x^{17} + x^{16} + x^{10} + x^{7} + x^{6} + x^{5} + x^{3} + x^{2} + x^{1} + 1$$

This short spreading sequence is called the pilot PN sequence, and it is based on the following characteristic polynomials used in Quadrature Modulation

$$P_I(x) = x^{15} + x^{13} + x^9 + x^8 + x^7 + x^5 + 1$$

for the in-phase (I) modulation and

$$P_Q(x) = x^{15} + x^{12} + x^{11} + x^{10} + x^6 + x^5 + x^4 + x^3 + 1$$

for the quadrature (Q) modulation.

Based on the characteristic polynomials, the pilot PN sequences $i(n)$ and $q(n)$ are generated by the following linear recursions:

$$i(n) = i(n\text{-}15) \oplus i(n\text{-}10) \oplus i(n\text{-}8) \oplus i(n\text{-}7) \oplus i(n\text{-}6) \oplus i(n\text{-}2)$$

and,

$$q(n) = q(n\text{-}15) \oplus q(n\text{-}13) \oplus q(n\text{-}11) \oplus q(n\text{-}10) \oplus q(n\text{-}9) \oplus q(n\text{-}5) \oplus q(n\text{-}4) \oplus q(n\text{-}3)$$

Power Control

- To minimize the average BER for each user, IS-95 strives to force each user to provide the same power level at the base station receiver.
- both signal and interference are continually varying, power control updates are sent by the base station every 1.25 ms.
- Power control commands are sent to each subscriber unit on the forward control subchannel which instruct the mobile to raise or lower its transmitted power in 1 dB steps.
- Power control bits are transmitted by using puncturing techniques.
- During a 1.25 ms period, twenty-four data symbols are transmitted, and IS-95 specifies sixteen possible power control group positions for the power control bit.
- Each position corresponds to one of the first sixteen modulation symbols.
- Twenty-four bits from the long code decimator are used for data scrambling in a period of 1.25 ms

Orthogonal Covering

- Each traffic channel transmitted on the forward CDMA channel is spread with a Walsh function at a fixed chip rate of 1.2288 Mbps.
- The Walsh functions comprise of sixty-four binary sequences, each of length 64, which are completely orthogonal to each other and provide orthogonal channelization for all users on the forward link.
- The Walsh functions comprise of sixty-four binary sequences, each of length 64, which are completely orthogonal to each other and provide orthogonal channelization for all users on the forward link.
- A user that is spread using Walsh function n is assigned channel number n (n = 0 to 63).
- The Walsh sequence repeats every 52.083 µs, which is equal to one coded data symbol. In other words, each data symbol is spread by 64 Walsh chips.

Walsh Hadamard Matrix

The 64 by 64 Walsh function matrix (also called a Hadamard matrix) is generated by the following recursive procedure:

$$H_1 = 0 \qquad H_2 = \begin{matrix} 0 & 0 \\ 0 & 1 \end{matrix}$$

$$H_4 = \begin{matrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{matrix} \qquad H_{2N} = \begin{matrix} H_N & H_N \\ H_N & \overline{H_N} \end{matrix}$$

, where N is a power of 2.

Quadrature Modulation

- After the orthogonal covering, symbols are spread in quadrature.
- A short binary spreading sequence, with a period of chips, is used for easy acquisition and synchronization at each mobile receiver and is used for modulation.
- This short spreading sequence is called the pilot PN sequence, and it is based on the characteristic polynomials P_I & P_Q
- Based on the characteristic polynomials, the pilot PN sequences i(n) and q(n) are generated by linear recursions.
- The initial state of both land Q pilot PN sequences is defined as the state in which the output of the pilot PN sequence generator is the first '1' output following fifteen consecutive '0' outputs.
- The chip rates for the pilot PN sequences are 1.2288 Mbps. The binary I and Q outputs of the quadrature spreading are mapped into phase accordingly.

Forward CDMA Channel I and Q Mapping

I	*Q*	Phase
0	0	$\pi/4$
1	0	$3\pi/4$
1	1	$-3\pi/4$
0	1	$-\pi/4$

IS-95 Reverse Channel

The block diagram of IS-95 reverse channel is shown in Figure 10.

- User data on the reverse channel are grouped into 20 ms frames.
- All data transmitted on the reverse channel are convolutionally encoded, block interleaved, modulated by a 64-ary orthogonal modulation, and spread prior to transmission.
- The speech or user data rate in the reverse channel may be sent at 9600, 4800, 2400, or 1200 bps.
- The reverse CDMA channels are made up of access channels (AC) and reverse traffic channels (RTC).
- Both share the same frequency assignment, and each Traffic/Access channel is identified by a distinct user long code.
- The Reverse CDMA channel may contain a maximum of 32 ACs per supported paging channel. While the RTC operates on a variable data rate, the AC works at a fixed data rate of 4800 bps.

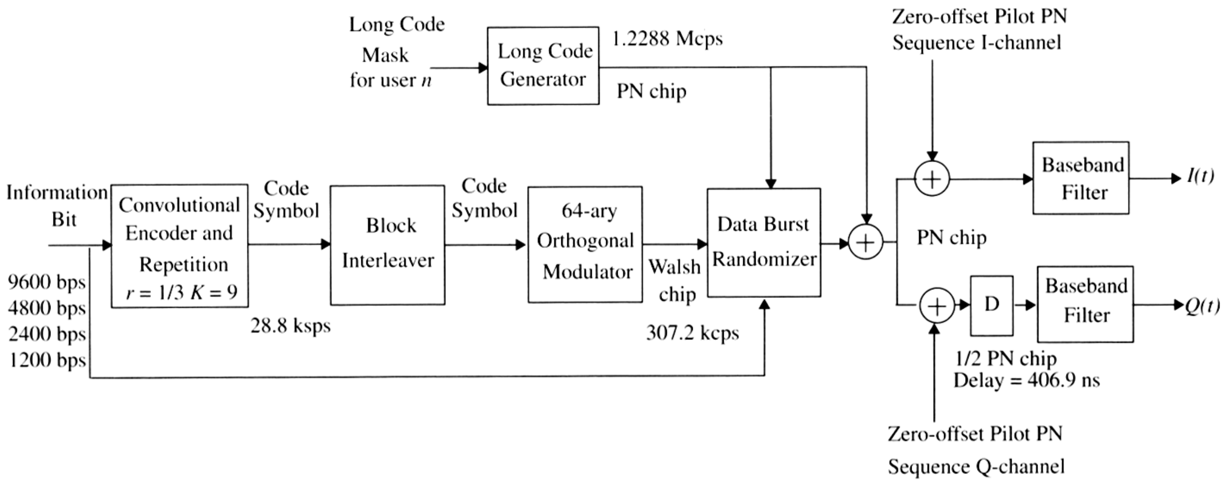


Figure 10: IS-95 Reverse channel modulation for a single user

Convolutional Encoder and Symbol Repetition

- The convolutional coder used in the reverse traffic channel is rate 1/3 and constraint length is 9.
- The three generator vectors g0, g1, and g2 are 557 (octal), 663 (octal), and 771 (octal), respectively.
- Coded bits after the convolutional encoder are repeated before interleaving when the data rate is less than 9600 bps. This is identical to the method used on the forward channel.
- After repetition, the symbol rate out of the coder is fixed at 28,800 bps.

Block interleaver

- It is performed following convolutional encoding and repetition.
- The block interleaver spans 20 ms, and is an array with 32 rows and 18 columns.

Orthogonal Modulation

- A 64-ary orthogonal modulation is used for the reverse CDMA channel.
- One of sixty-four possible Walsh functions is transmitted for each group of six coded bits.
- Within a Walsh function, sixty-four Walsh chips are transmitted.
- Walsh function is selected according to the following formula:
  Walsh function number = $c_0 + 2c_1 + 4c_2 + 5c_3 + 16c_4 + 32c_5$,
  where c_5 represents the last coded bit and c_0 represents the first coded bit of each group of six coded symbols that are used to select a Walsh function. Walsh chips are transmitted at a rate of 307.2 kcps
  28.8 kbps x (64 Walsh chips) / (6 coded bits) = 307.2 kbps
- Walsh functions are used for different purposes on the forward and reverse channels. On the forward channel, Walsh functions are used for spreading to denote a particular user channel, while on the reverse channel, Walsh functions are used for data modulation.

Data burst randomizer

- Reduce interference from other mobile stations
- Using long code mask to smooth data out over 20 ms frame
- The data burst randomizer generates a masking pattern 0"s and 1's that randomly masks the redundant data generated by the code repetition process.
- A block of 14 bits taken from the long code determines the masking pattern.
- The last 14 bits of the long code used for spreading in the second to last power control group of the previous frame are used to determine the random mask for the gating.

Direct sequence spreading

- Long code unique to mobile XOR ed with output of randomizer
- 1.2288-Mbps final data stream
- Modulated using orthogonal QPSK modulation scheme
- Differs from forward channel in use of delay element in modulator to produce orthogonality.
- Each Walsh chip is spread by four long code PN chips.
- Reverse channel orthogonality of spreading codes not guaranteed

Quadrature Modulation

- Prior to transmission, the reverse traffic channel is spread by I and Q channel
- pilot PN sequences are identical to those used in the forward CDMA channel process. These pilot sequences are used for synchronization purpose.
- The reverse link modulation is offset quadrature phase shift keying (OQPSK).
- The data spread by the Q pilot PN sequence is delayed by half a chip (406.901 ns) with respect to the data spread by the I pilot PM sequence.
- This delay is used for improved spectral shaping and synchronization

Multi carrier modulation

- The basic idea of multicarrier modulation is to divide the transmitted bit stream into many different substreams and send these over many different subchannels.
- Typically the sub channels are orthogonal under ideal propagation conditions. The data rate on each of the subchannels is much less than the total data rate, and the corresponding subchannel bandwidth is much less than the total system bandwidth.
- The number of substreams is chosen to ensure that each subchannel has a bandwidth less than the coherence bandwidth of the channel, so the subchannels experience relatively flat fading. Thus, the intersymbol interference on each subchannel is small.
- The subchannels in multicarrier modulation need not be contiguous, so a large continuous block of spectrum is not needed for high-rate multicarrier communications.
- Moreover, multicarrier modulation is efficiently implemented digitally.
- In this discrete implementation, called orthogonal frequency division multiplexing (OFDM), the ISI can be completely eliminated through the use of a cyclic prefix

Multi carrier Transmitter

The block diagram of multi carrier modulation is shown in Figure 11.

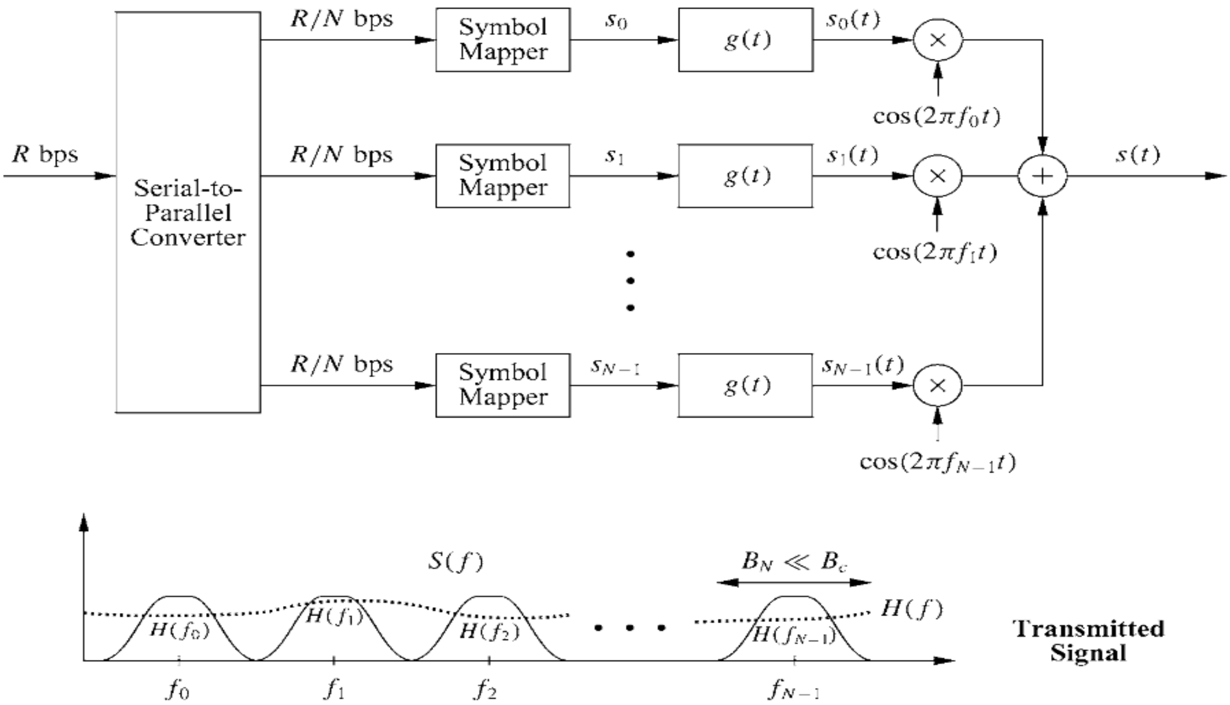


Figure 11: Multi carrier Transmitter

- The bit stream is divided into N sub-streams via a serial-to-parallel converter.
- The nth substream is linearly modulated (typically via QAM or PSK) relative to the subcarrier frequency f n and occupies bandwidth B_N .
- We assume coherent demodulation of the subcarriers so the subcarrier phase is neglected in our analysis.
- If we assume raised cosine pulses for g(t) we get a symbol time
  $T_N = (1 + \beta)/B_N$
  for each substream, where β is the rolloff factor of the pulse shape.

- The modulated signals associated with all the subchannels are summed together to form the transmitted signal, given as

$$s(t) = \sum_{i=0}^{N-1} s_i g(t) \cos(2\pi f_i t + \phi_i),$$

where s_i is the complex symbol associated with the ith subcarrier and φ_i is the phase offset of the ith carrier.

- For nonoverlapping subchannels we set $f_i = f_0 + i(B_N)$, $i = 0, \ldots, N-1$.
- The substreams then occupy orthogonal subchannels with bandwidth B_N , yielding a total bandwidth $NB_N = B$ and data rate $NR_N \approx R$.
- Thus, this form of multicarrier modulation does not change the data rate or signal bandwidth relative to the original system, but almost completely eliminates ISI for $B_N << B_c$

Multi carrier receiver

The receiver for this multicarrier modulation is shown in Figure 12.
Each substream is passed through a narrowband filter (to remove the other substreams), demodulated, and combined via a parallel-to-serial converter to form the original data stream. Note that the ith subchannel will be affected by flat fading corresponding to a channel gain
$\alpha_i = H(f_i)$.

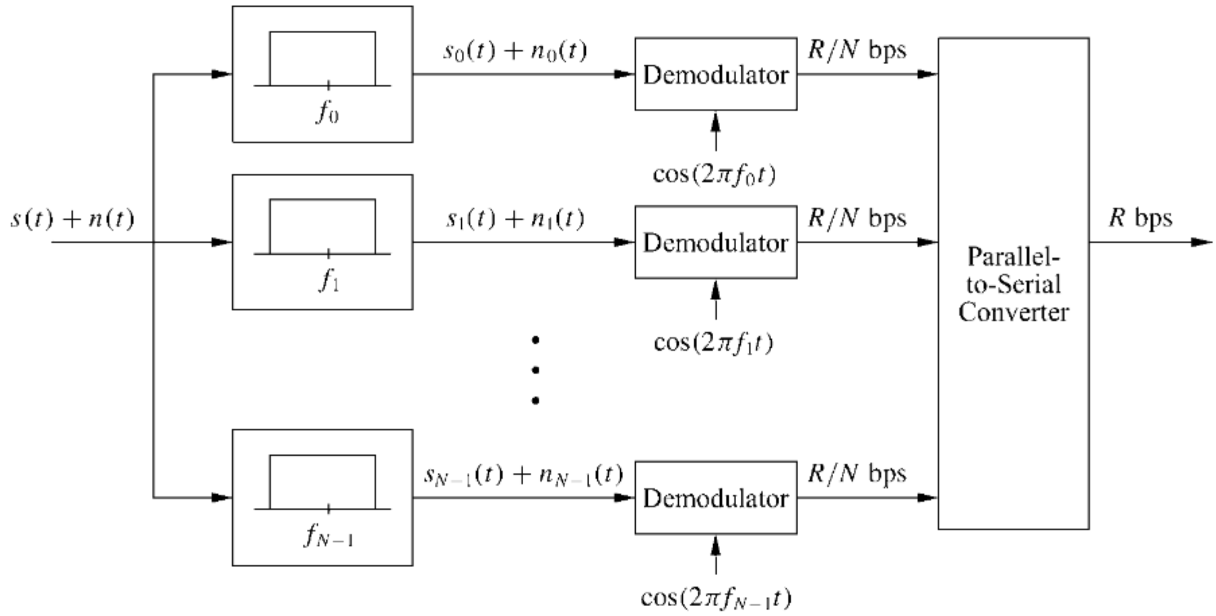


Figure 12. Multi Carrier receiver

Orthogonal Frequency Division Mutiplexing (OFDM) Transmitter

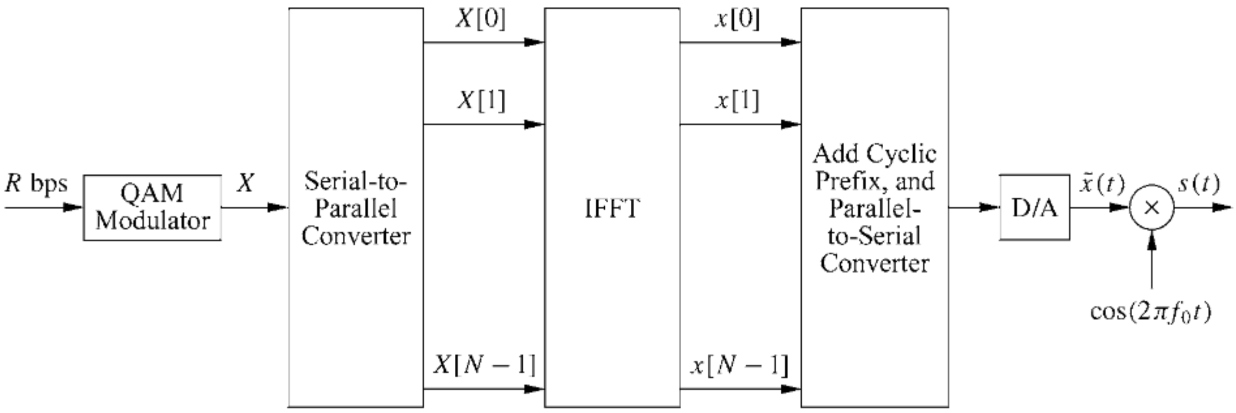


Figure 13: Block diagram of OFDM Transmitter

The OFDM implementation of multicarrier modulation is shown in Figure 13.

- The input data stream is modulated by a QAM modulator, resulting in a complex symbol stream X[0], X[1], . . . , X[N − 1].

- This symbol stream is passed through a serial-to-parallel converter, whose output is a set of N parallel QAM symbols X[0], . . . , X[N − 1] corresponding to the symbols transmitted over each of the subcarriers.
- Thus, the N symbols output from the serial-to-parallel converter are the discrete frequency components of the OFDM modulator output s(t).
- In order to generate s(t), the frequency components are converted into time samples by performing an inverse DFT on these N symbols, which is efficiently implemented using the IFFT algorithm.
- The IFFT yields the OFDM symbol consisting of the sequence
  x[n] = x[0], . . . , x[N − 1] of length N, where

$$x[n] = \frac{1}{\sqrt{N}} \sum_{i=0}^{N-1} X[i] e^{j2\pi ni/N}, \quad 0 \le n \le N-1.$$

- This sequence corresponds to samples of the multicarrier signal: the multicarrier signal consists of linearly modulated subchannels,
- The right side of above equation corresponds to samples of a sum of QAM symbols X[i] each modulated by the carrier $e^{j2\pi it/N}$, i = 0, . . . , N −1.
- The cyclic prefix is then added to the OFDM symbol, and the resulting time samples
- $\tilde{x}$[n] = $\tilde{x}$ [−μ], . . . , $\tilde{x}$ [N −1] = x[N −μ], . . . , x[0], . . . , x[N −1] are ordered by the parallel-to-serial converter and passed through a D/A converter, resulting in the baseband OFDM signal $\tilde{x}$ (t), which is then upconverted to frequency f_0.

OFDM Receiver

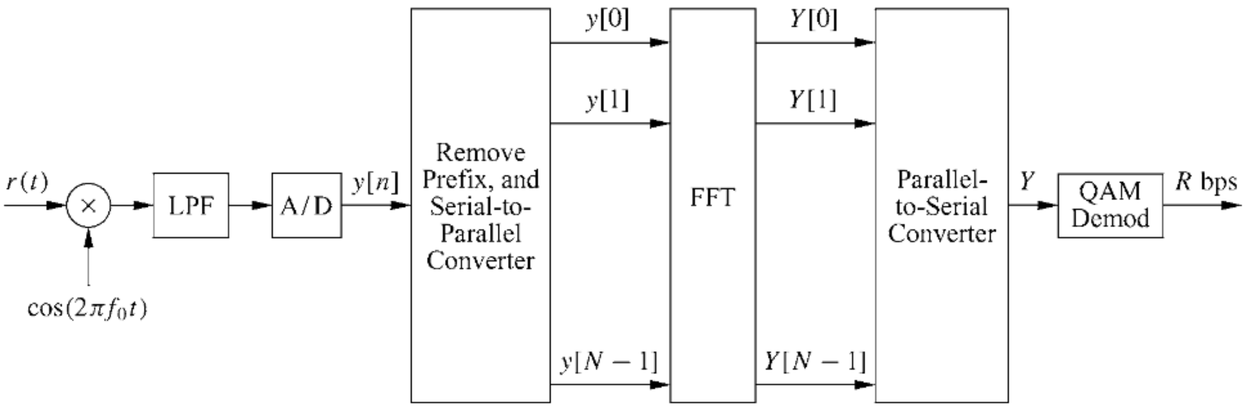


Figure 14: Block diagram of OFDM Receiver

The OFDM implementation of multi carrier receiver is shown in Figure 14.

- The transmitted signal is filtered by the channel impulse response and corrupted by additive noise, resulting in the received signal r(t).
- This signal is down converted to baseband and filtered to remove the high-frequency components. The A / D converter samples the resulting signal to obtain
- y[n] = $\tilde{x}$[n] ∗ h[n] + ν[n], −μ ≤ n ≤ N −1, where h[n] is the discrete-time equivalent low pass impulse response of the channel. The prefix of y[n] consisting of the first μ samples is then removed.
- This results in N time samples whose DFT in the absence of noise is Y [i] = H [i]X[i].
- These time samples are serial-to-parallel converted and passed through an FFT.

- This results in scaled versions of the original symbols H [i]X[i], where H [i] = H(f_i) is the flat fading channel gain associated with the ith subchannel.
- The FFT output is parallel to-serial converted and passed through a QAM demodulator to recover the original data
- The OFDM system effectively decomposes the wideband channel into a set of narrowband orthogonal subchannels with a different QAM symbol sent over each subchannel.
- Knowledge of the channel gains H [i], i = 0, . . . , N − 1, is not needed for this decomposition, in the same way that a continuous-time channel with frequency response H(f) can be divided into orthogonal subchannels without knowledge of H(f) by splitting the total signal bandwidth into nonoverlapping subbands.
- The demodulator can use the channel gains to recover the original QAM symbols by dividing out these gains: X[i] = Y [i]/H [i].
- This process is called frequency equalization

Cyclic prefix

- Consider a channel input sequence x[n] = x[0], . . . , x[N − 1] of length N and a discrete time channel with finite impulse response (FIR) h[n] = h[0], . . . , h[μ] of length μ + 1 = T_m /T_s , where T_m is the channel delay spread and T_s the sampling time associated with the discrete time sequence.
- The cyclic prefix for x[n] is defined as {x[N − μ], . . . , x[N − 1]},it consists of the last μ values of the x[n] sequence.
- For each input sequence of length N, these last μ samples are appended to the beginning of the sequence. This yields a new sequence $\tilde{x}$ [n], −μ ≤ n ≤ N − 1, of length N + μ, where $\tilde{x}$ [−μ], . . . , $\tilde{x}$ [N − 1] = x[N − μ], . . . , x[N − 1], x[0], . . . , x[N − 1], as shown in Fig. 14
- Note that with this definition, $\tilde{x}$ [n] = x[n]N for −μ ≤ n ≤ N − 1, which implies that $\tilde{x}$ [n − k] = x[n − k]N for −μ ≤ n − k ≤ N − 1.

Suppose $\tilde{x}$ [n] is input to a discrete-time channel with impulse response h[n].
The channel output y[n], 0 ≤ n ≤ N − 1, is then

$$\begin{aligned} y[n] &= \tilde{x}[n] * h[n] \\ &= \sum_{k=0}^{\mu} h[k]\tilde{x}[n-k] \\ &= \sum_{k=0}^{\mu} h[k]x[n-k]_N \\ &= x[n] \circledast h[n], \end{aligned}$$

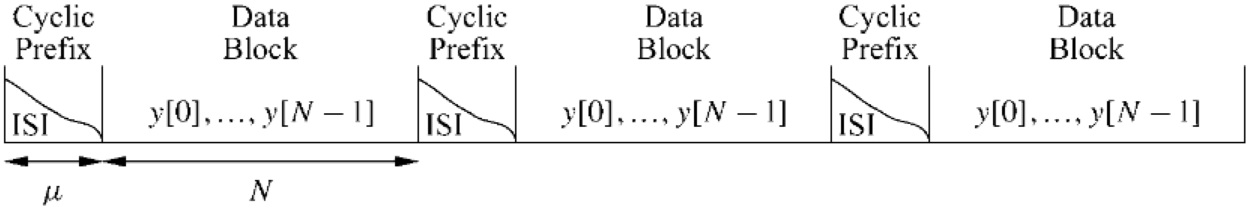


Figure 15: ISI between data blocks in channel output.

where the third equality follows from the fact that, for 0 ≤ k ≤ μ, [n − k] = x[n − k]N for 0 ≤ n ≤ N − 1.

- **Thus, by appending a cyclic prefix to the channel input, the linear convolution associated with the channel impulse response y[n] for $0 \le n \le N - 1$ becomes a circular convolution.**
- **The cyclic prefix serves to eliminate ISI between the data blocks**, because the first μ samples of the channel output affected by this ISI can be discarded without any loss relative to the original information sequence.
- In continuous time this is equivalent to using a guard band of duration T_m (the channel delay spread) after every block of N symbols of duration NT_s in order to eliminate the ISI between these data blocks.
- The benefits of adding a cyclic prefix come at a cost. Since μ symbols are added to the input data blocks, there is an overhead of μ/N and a resulting data-rate reduction of N/(μ + N).
- The transmit power associated with sending the cyclic prefix is also wasted because this prefix consists of redundant data.
- It is clear from Figure 15, that any prefix of length μ appended to input blocks of size N eliminates ISI between data blocks if the first μ samples of the block are discarded.

An alternative to using the cyclic prefix is to use a prefix consisting of all zero symbols. In this case the OFDM symbol consisting of x[n], $0 \le n \le N - 1$, is preceded by μ null samples, as shown in Figure 16.

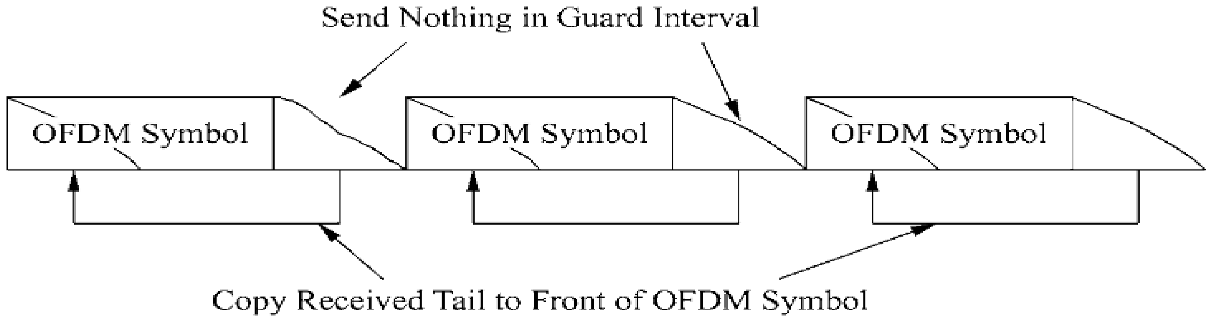


Figure 16: Creating a circular channel with an all-zero prefix

- At receiver, tail of the ISI associated with the end of a given OFDM symbol is added back in to the beginning of the symbol, which re-creates the effect of a cyclic prefix, so the rest of the OFDM system functions as usual.
- This zero prefix reduces the transmit power relative to a cyclic prefix by N/(μ + N), since the prefix does not require any transmit power.
- However, the noise from the received tail is added back into the beginning of the symbol, which increases the noise power by (N + μ)/N.

Modern Antennas

Automotive antennas

- To create smoother and safer journey, connected vehicles utilize high speed mobile connection to interact with other vehicles. There comes automotive antennas to serve the purpose.
- This antenna will integrate the wireless communication services in a vehicle.
- They pave the way for many applications like vehicle to vehicle (V2V) communication, vehicle to Infrastructure (V2I) communication, Intelligent Transport System, Traffic management and collision avoidance.

- For driver information, modern navigation system not only helps to find efficient route but also to give overview of traffic situation.

Challenges

- To design a compact antenna
- To overcome the mutual coupling effect and interference from the electronic devices near by the vehicle.
- To identify the location in vehicle where the antenna to be placed.
- The placement of antenna should not affect the aesthetic appearance of car.

❖ **Wearable antennas**

- The demand for wearable electronics and related technologies have grown tremendously in recent years. Some of the key developments that accelerated this growth are miniaturization of wireless devices, advent of high-speed wireless networks, availability of ultra-compact, low-power SoCs and ever-evolving battery technologies.
- Wearable electronics find numerous applications these days, and most of these applications use different types of antennas to sense, fetch, and exchange data wirelessly to and from a host device or an IoT gateway.
- Wearable Antennas are essentially any antenna that is specifically designed to function while being worn. Examples include smart watches, glasses (such as Google Glass which has WiFi and GPS antennas).
- Wearable antennas are designed to function while being worn. These antennas are commonly used in wearable wireless communication and bio-medical RF systems. Wearable antennas are used within the context of Wireless Body Area Networks (WBAN).
- In a WBAN, antenna is the key component that supports wireless communication, which include in-body communication, on-body communication and off-body communication.
- A WBAN connects sensors, actuators and IoT nodes on human body, or on clothes or under the skin, establishing a wireless communication channel.
- Wearable antennas can be employed on people of all ages, athletes and patients for a continuous monitoring of vital signs, oxygen level (Oximetry) and stress level, among others.

❖ **Reflectarray antenna**

- It is a novel type of high gain antenna which consists of an array of unit cells, consisting of either a flat or a slightly curved reflecting surface illuminated by a feeding antenna. The feeding antenna is usually a horn.

Advantages of Reflectarray Antenna

- Simple structure and Low cost
- Ease of fabrication
- Shared Aperture
- Dual or multiple beam coverage
- No feed network loses
- Long distance communication
- It is used in satellite communication, Radars, Astronomical observations etc.

❖ Transmitarray antenna

- A high gain can be realized using two approaches:
  - One is based on the optic theory that manipulates the geometrical curvature of antenna surface to focus the radiation beam.
  - The other is the antenna array theory that controls the interference of elements radiation appropriately.
- Representations for the first approach are the parabolic reflectors and lens antennas, and examples of the second approach include waveguide-slot arrays and printed microstrip antenna arrays.
- As an emerging concept, the transmitarray antenna combines the favorable features of optic theory and antenna array techniques, leading to a low profile conformal design with high radiation efficiency and versatile radiation performance
- A transmitarray antenna consists of an illuminating feed source and a thin transmitting surface.
- The feed source is located on an equivalent focal point. On the transmitting surface, there is an array of antenna elements.
- The transmission coefficients of these elements are individually designed to convert the spherical phase front from the feed to a planar phase front. As a result, a focused radiation beam can be achieved with a high gain.
- Transmitarray antennas have a great potential in many applications such as earth remote sensing, wireless communications, spatial power combining for high power applications, THz images and sensors, and solar energy concentrator.

❖ Antennas for Non-destructive Testing

- **Non-destructive testing (NDT)**
  - o Determining the characteristics of the target system under test without hindering or compromising its utility or usefulness
- **Traditional methods of NDT**
  - o Ultrasound, eddy current, stereography, magnetic particle testing, dye penetrant, visual testing, and radiography.
- **Application domains**
  - o Infrastructure, oil gas pipelines, aerospace maintenance repair and overhaul (MRO) for detection and imaging of defects and ensuring the structural health of the target under test.
- **Why UWB for NDT Applications?**
  - o It can operate in both low and high frequency range giving the unique features like:
  - o Ability to penetrate obstacles, Higher precision range and multipath resolving capacity, Low EM radiation of -41.3 dB, Low energy consumption
- **Recent research works**
  - o Detection of Moisture Content and Deterioration in Concrete
  - o A Non-Invasive Bone Fracture Monitoring Analysis using an UHF Antenna
  - o Design of Ultra-Wide Band Antenna for breast tumor detect