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NEAR EAST UNIVERSITY

FACULITY OF ENGINEERING

Department of Electrical and Electronic Engineering

SATELITE DATA PROTOCOLES, ENCODING AND PERFORMANCE ISSUES ANALYSIS

Graduation Project
EE-400

Student: Salman Sultan (980814)

Supervisor: Mr. Izzet Agoren

Nicosia - 2004

ACKNOWLEDGEMENTS

In the name of Allah whose the most gracious and most merciful.

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Abstract

The project examined an extensible software framework for the purpose of data performance of the DVB-T standard. In order to examine the software simulations were performed and compared with expected channels results. It was determined that the channel we use has the better performance then the other. It was concluded that this channel can be used by research depts. And companies to develop and test new applications for DVB-T systems before going through the expensive prototyping process.

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INTRODUCTION

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The project is aimed to provide analysis and result of the DVB-T model. The project consists of the introduction, four chapters and the conclusion.

The first chapter gives the brief explanation of Satellite data protocol and give the introduction of modulation.

Second chapter gives the background and working behavior of each topic we discuss in first chapter.

Third chapter is an experiment chapter .Which presents the introduction of channels and basic Parameters used in experiment.

Fourth Chapter provides the result and compares the result of channel. The conclusion present important result obtained and practical realization of the DVB-T.

INTRODUCTION TO DATA PROTOCOLES

1.1 Asynchronous Transfer Mode (ATM) Protocol

Many designers of satellite systems are thinking about the application of the asynchronous transfer mode (ATM) protocol The ATM protocol transmits data that have been placed in cells of a constant length (53 bytes). The ATM guarantees data transmission at a rate ranging between 2 Mb/s and 2.4 Gb/s. The protocol acts on the principle that a virtual channel should be set up between two points whenever such a need appears. This is what makes the ATM protocol different from the TCP/IP protocol, in which messages are transmitted in packet form, where each packet may reach the recipient via a different route. The ATM protocol enables data transmission through various media. However, taking into account the header of the cell (cell-tax) which takes 5 bytes, the application of the ATM protocol may appear not to be so cost-effective when the rate of transmission is low, and the capacity of the link (e.g., in two-way modem channels), becomes a basic limitation.

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1.4 Application of Data Protocol

In ETSI standards there are three kinds of standers namely DVB-T (DIGITAL VIDEO BROAD CASTING-TERRESTAL), DVB-S (DIGITAL VIDEO BROADCASTIN SATELLITE) and DVB-C (DIGITAL VIDEO BROADCASTING CABLE).

1.4.1 Digital Video Broad Casting-Terrestal (DVB-T)

The system is defined as the functional block of equipment performing the adaptation of the base band TV signals from the output of the MPEG-2 transport multiplexer, to the terrestrial channel characteristics. The following processes shall be applied to the data stream.

- Transport multiplex adaptation and randomization for energy dispersal;
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This digital frequency modulation technique is primarily used for sending data downstream over a coaxial cable network. 64QAM is very efficient, supporting up to 28-mbps peak transfer rates over a single 6-MHz channel. But 64QAM's susceptibility to interfering signals makes it ill suited to noisy upstream transmissions (from the cable subscriber to the Internet).

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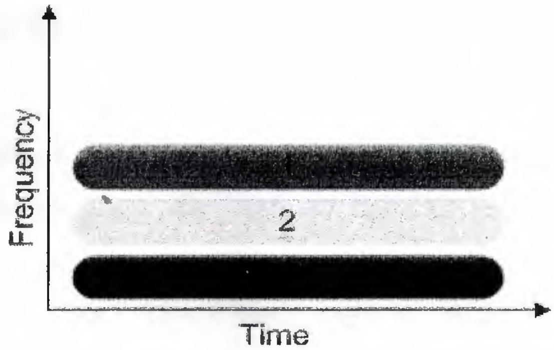
There are 4 main multiple access Schemes which are as follow.

1.6.1 Frequency Division Multiple Access

Frequency Division Multiple Access. A unique frequency slot is assigned to each user for the duration of their call. The number of users within a cell is determined by the number of distinct frequency slots available. In the figure no. 1.6.1, 3 users are each allocated a unique frequency band that only they may use. More than one user may transmit on the same channel at once, leading to possible cross-talk (non-linearities in the channel).



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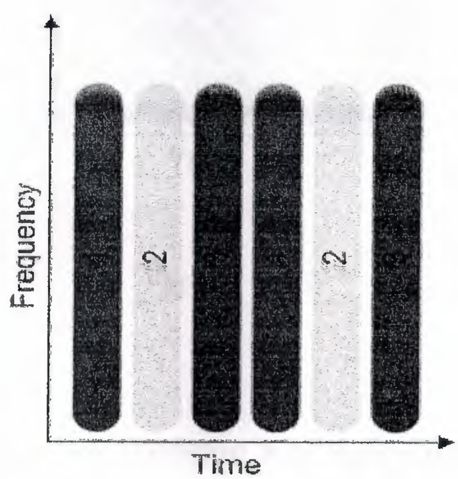
Time Division Multiple Access. The frequency band is not partitioned as in FDMA, but only one user can access the channel at any specific time. Each user is assigned a distinct time slot to access the channel as can be seen in the figure no. 1.6.2. The same 3 users are now allocated a unique time slot, and each user may only access the entire channel in their unique slot. It is essential that there is perfect synchronization for the system to function adequately.



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Code Division Multiple Access. A spread spectrum technique, which employs the use of spreading codes to allow the users to transmit simultaneously at the same frequency. Each users signal occupies the entire bandwidth. The figure below is a crude illustration of this idea. The 3 same users as pictured above in TDMA and FDMA now have the use of the entire bandwidth. The boxes of colour are not fixed and are used to show the users using the entire frequency, what the illustration should look like is a mixture of the 3 colours overlapping. The spreading code is unique to the user and the same code is used at the receiver to decode the signal.

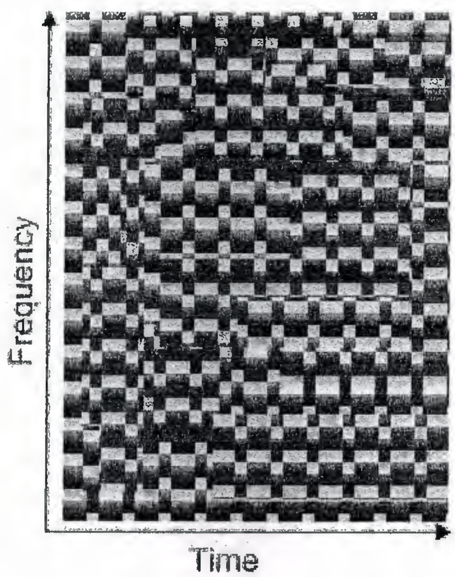

Fig No. 1.6.3

1.6.4 Orthogonal Frequency Division Multiplexing

Frequency division multiplexing (FDM) is a technology that transmits multiple signals simultaneously over a single transmission path, such as a cable or wireless system. Each signal travels within its own unique frequency range (carrier), which is modulated by the data (text, voice, video, etc.).

Orthogonal FDM's (OFDM) spread spectrum technique distributes the data over a large number of carriers that are spaced apart at precise frequencies. This spacing provides the "orthogonality" in this technique which prevents the demodulators from seeing frequencies other than their own. The benefits of OFDM are high spectral efficiency, resiliency to RF interference, and lower multi-path distortion. This is useful because in a typical terrestrial broadcasting scenario there are multipath-channels (i.e. the transmitted signal arrives at the receiver using various paths of different length). Since multiple versions of the signal interfere with each other (inter symbol interference (ISI)) it becomes very hard to extract the original information.

Chapter 2
Back ground

2.1 The Background of Asynchronous Transfer Mode (ATM)

The Asynchronous Transfer Mode (ATM) was born out of standardization efforts for Broadband ISDN which began in the CCITT in the mid 1980s. It was originally intimately bound up with the emerging Synchronous Digital Hierarchy (SDH) standards, and was conceived as a way in which arbitrary-bandwidth communication channels could be provided within a multiplexing hierarchy consisting of a defined set of fixed-bandwidth channels.

The basic principles of ATM as put forward by CCITT in Recommendation I.150 are:

- ATM is considered as a specific packet oriented transfer mode based on fixed length cells. Each cell consists of an information field and a header, which is mainly used to determine the virtual channel and to perform the appropriate routing. Cell sequence integrity is preserved per virtual channel.
- ATM is connection-oriented. The header values are assigned to each section of a connection for the complete duration of the connection. Signaling and user information are carried on separate virtual channels.
- The information field of ATM cells is carried transparently through the network. No processing like error control is performed on it inside the network.
- All services (voice, video, data) can be transported via ATM, including connectionless services. To accommodate various services an adaptation function is provided to fit information of all services into ATM cells and to provide service specific functions (e.g. clock recovery, cell loss recovery,).

2.1.1 ATM Cell Format

ATM transmits switches and multiplexes information in fixed-length cells. The length of a cell is 53 bytes, consisting of a 5-byte cell header and 48 bytes of data.

Fig. 2.1 ATM Cell

2.1.2 Header Format

The ATM header contains information about destination, type and priority of the cell. The Generic Flow Control (GFC) field allows a multiplexer to control the rate of an ATM terminal. The GFC field is only available at the User-to-Network Interface (UNI). At the Network-to-Network Interface (NNI) these bits belong to the Virtual Path Identifier (VPI).

The Virtual Path Identifier (VPI) and the Virtual Channel Identifier (VCI) hold the locally valid relative address of the destination. These fields may be changed within an ATM switch.

The Payload Type (PT) marks whether the cell carries user data, signaling data or maintenance information.

The Cell Loss Priority (CLP) bit indicates which cells should be discarded first in the case of congestion.

Finally, the Header Error Control (HEC) field is to perform a CRC check on the header data. Only the header is error checked in the ATM layer. Error check for the user data is left to higher layer protocols and is performed on an end-to-end base.

<table>
<tr><th></th><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td>Byte 1</td><td colspan="4">GFC</td><td colspan="4">VPI</td></tr>
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Table 2.1 ATM cell header format (UNI)

Abbreviations	
GFC	Generic Flow Control
VPI	Virtual Path Identifier
VCI	Virtual Channel Identifier
PTI	Payload Type Identifier
CLP	Cell Loss Priority
HEC	Header Error Control

Table 2.2 ATM cell header abbreviations

2.1.3 Quality of Service (QoS)

ATM Networks are thought to transmit data with varying characteristics. Different applications need various Qualities of Service (QoS). Some applications like telephony may be very sensitive to delay, but rather insensitive to loss, whereas others like compressed video are quite sensitive to loss.

The ATM Forum specified several Quality of Service (QoS) categories:

- CBR (Constant Bit Rate)
- Rt-VBR (real-time Variable Bit Rate)
- Nrt-VBR (non-real-time Variable Bit Rate)
- ABR (Available Bit Rate)
- UBR (Unspecified Bit Rate)

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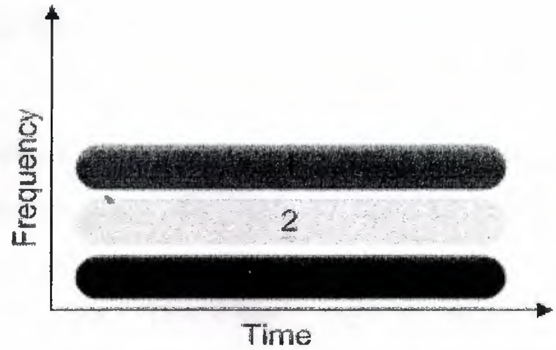

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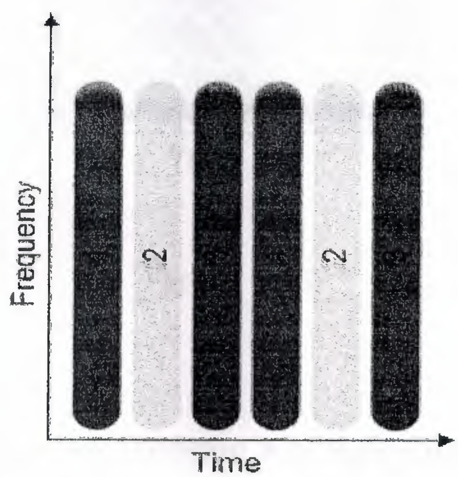

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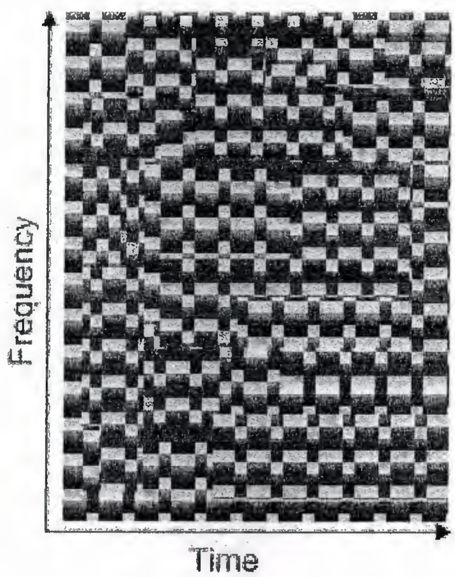

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Chapter 2
Back ground

2.1 The Background of Asynchronous Transfer Mode (ATM)

The Asynchronous Transfer Mode (ATM) was born out of standardization efforts for Broadband ISDN which began in the CCITT in the mid 1980s. It was originally intimately bound up with the emerging Synchronous Digital Hierarchy (SDH) standards, and was conceived as a way in which arbitrary-bandwidth communication channels could be provided within a multiplexing hierarchy consisting of a defined set of fixed-bandwidth channels.

The basic principles of ATM as put forward by CCITT in Recommendation I.150 are:

- ATM is considered as a specific packet oriented transfer mode based on fixed length cells. Each cell consists of an information field and a header, which is mainly used to determine the virtual channel and to perform the appropriate routing. Cell sequence integrity is preserved per virtual channel.
- ATM is connection-oriented. The header values are assigned to each section of a connection for the complete duration of the connection. Signaling and user information are carried on separate virtual channels.
- The information field of ATM cells is carried transparently through the network. No processing like error control is performed on it inside the network.
- All services (voice, video, data) can be transported via ATM, including connectionless services. To accommodate various services an adaptation function is provided to fit information of all services into ATM cells and to provide service specific functions (e.g. clock recovery, cell loss recovery,).

2.1.1 ATM Cell Format

ATM transmits switches and multiplexes information in fixed-length cells. The length of a cell is 53 bytes, consisting of a 5-byte cell header and 48 bytes of data.

Fig. 2.1 ATM Cell

2.1.2 Header Format

The ATM header contains information about destination, type and priority of the cell. The Generic Flow Control (GFC) field allows a multiplexer to control the rate of an ATM terminal. The GFC field is only available at the User-to-Network Interface (UNI). At the Network-to-Network Interface (NNI) these bits belong to the Virtual Path Identifier (VPI).

The Virtual Path Identifier (VPI) and the Virtual Channel Identifier (VCI) hold the locally valid relative address of the destination. These fields may be changed within an ATM switch.

The Payload Type (PT) marks whether the cell carries user data, signaling data or maintenance information.

The Cell Loss Priority (CLP) bit indicates which cells should be discarded first in the case of congestion.

Finally, the Header Error Control (HEC) field is to perform a CRC check on the header data. Only the header is error checked in the ATM layer. Error check for the user data is left to higher layer protocols and is performed on an end-to-end base.

	8	7	6	5	4	3	2	1
Byte 1	GFC				VPI			
Byte 2	VPI				VCI			
Byte 3	VCI							
Byte 4	VCI				PTI			CLP
Byte 5	HEC							

Table 2.1 ATM cell header format (UNI)

Abbreviations	
GFC	Generic Flow Control
VPI	Virtual Path Identifier
VCI	Virtual Channel Identifier
PTI	Payload Type Identifier
CLP	Cell Loss Priority
HEC	Header Error Control

Table 2.2 ATM cell header abbreviations

2.1.3 Quality of Service (QoS)

ATM Networks are thought to transmit data with varying characteristics. Different applications need various Qualities of Service (QoS). Some applications like telephony may be very sensitive to delay, but rather insensitive to loss, whereas others like compressed video are quite sensitive to loss.

The ATM Forum specified several Quality of Service (QoS) categories:

- CBR (Constant Bit Rate)
- Rt-VBR (real-time Variable Bit Rate)
- Nrt-VBR (non-real-time Variable Bit Rate)
- ABR (Available Bit Rate)
- UBR (Unspecified Bit Rate)

The following table shows, which are the negotiated parameters for any QoS category.

<table>
<tr><th></th><th>CBR</th><th>Rt-VBR</th><th>Nrt-VBR</th><th>UBR</th><th>ABR</th></tr>
<tr><td>Traffic Parameters:</td><td colspan="5"></td></tr>
<tr><td>PCR and CDVT</td><td colspan="5">Specified</td></tr>
<tr><td>SCR, MBS, CDVT</td><td>N/A</td><td colspan="2">Specified</td><td colspan="2">N/A</td></tr>
<tr><td>MCR</td><td colspan="4">N/A</td><td>Specified</td></tr>
<tr><td>QoS Parameters:</td><td colspan="5"></td></tr>
<tr><td>Peak-to-peak CDV</td><td colspan="2">Specified</td><td colspan="3">Unspecified</td></tr>
<tr><td>MaxCTD</td><td colspan="2">Specified</td><td colspan="3">Unspecified</td></tr>
<tr><td>CLR</td><td colspan="3">Specified</td><td>Unspecified</td><td>Network specific</td></tr>
<tr><td>Other Attributes:</td><td colspan="5"></td></tr>
<tr><td>Feedback</td><td colspan="4">Unspecified</td><td>Specified</td></tr>
</table>

Table 2.3 QoS parameters

Abbreviations	
CDV	Cell Delay Variation
CDVT	CDV Tolerance
CLR	Cell Loss Ratio
CTD	Cell Transfer Delay
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
PCR	Peak Cell Rate
SCR	Sustainable Cell Rate

Table 2.4 QoS abbreviations

2.1.4 Constant Bit Rate (CBR)

During a connection setup CBR reserves a constant amount of bandwidth. This service is conceived to support applications such as voice, video and circuit emulation, which require small delay variations (jitter). The source is allowed to send at the negotiated rate any time and for any duration. It may temporarily send at a lower rate as well.

2.1.5 Variable Bit Rate (VBR)

VBR negotiates the Peak Cell Rate (PCR), the Sustainable Cell Rate (SCR) and the Maximum Burst Size (MBS). VBR sources are bursty. Typical VBR sources are compressed voice and video. These applications require small delay variations (jitter). The VBR service is further divided in real-time VBR (Rt-VBR) and non-real-time VBR (Nrt-VBR). They are distinguished by the need for an upper bound delay (Max CTD).MaxCTD is provided by Rt-VBR, whereas for Nrt-VBR no delay bounds are applicable.

2.1.6 Available Bit Rate (ABR) and Unspecified Bit Rate (UBR)

ABR and UBR services should efficiently use the remaining bandwidth, which is dynamically changing in time because of VBR service. Both are supposed to transfer data without tight constraints on end-to-end delay and delay variation. Typical applications are computer communications, such as file transfers and e-mail.

UBR service provides no feedback mechanism. If the network is congested, UBR cells may be lost.

An ABR source gets feedback from the network. The network provides information about the available bandwidth and the state of congestion. The source's transmission rate is adjusted in function of this feedback information. This more efficient use of bandwidth alleviates congestion and cell loss. For ABR service, a guaranteed minimum bandwidth (MCR) is negotiated during the connection setup negotiations.

2.2 Background of Tcp/Ip

The Internet Protocol was developed to create a Network of Networks (the "Internet"). Individual machines are first connected to a LAN (Ethernet or Token Ring). TCP/IP shares the LAN with other uses (a Novell file server, Windows for Workgroups peer systems). One device provides the TCP/IP connection between the LAN and the rest of the world.

To insure that all types of systems from all vendors can communicate, TCP/IP is absolutely standardized on the LAN. However, larger networks based on long distances and phone lines are more volatile. In the US, many large corporations would wish to reuse large internal networks based on IBM's SNA. In Europe, the national phone companies traditionally standardize on X.25. However, the sudden explosion of high speed microprocessors, fiber optics, and digital phone systems has created a burst of new options: ISDN, frame relay, FDDI, Asynchronous Transfer Mode (ATM). New technologies arise and become obsolete within a few years. With cable TV and phone companies competing to build the National Information Superhighway, no single standard can govern citywide, nationwide, or worldwide communications.

The original design of TCP/IP as a Network of Networks fits nicely within the current technological uncertainty. TCP/IP data can be sent across a LAN, or it can be carried within an internal corporate SNA network, or it can piggyback on the cable TV service. Furthermore, machines connected to any of these networks can communicate to any other network through gateways supplied by the network vendor.

2.2.1 Addresses

Each technology has its own convention for transmitting messages between two machines within the same network. On a LAN, messages are sent between machines by supplying the six byte unique identifier (the "MAC" address). In an SNA network, every machine has Logical Units with their own network address. DECNET, AppleTalk, and Novell IPX all have a scheme for assigning numbers to each local network and to each workstation attached to the network.

On top of these local or vendor specific network addresses, TCP/IP assigns a unique number to every workstation in the world. This "IP number" is a four byte value that, by convention, is expressed by converting each byte into a decimal number (0 to 255) and separating the bytes with a period. For example, the PC Lube and Tune server is 130.132.59.234.

An organization begins by sending electronic mail to Hostmaster@INTERNIC.NET requesting assignment of a network number. It is still possible for almost anyone to get assignment of a number for a small "Class C" network in which the first three bytes identify the network and the last byte identifies the individual computer. The author followed this procedure and was assigned the numbers 192.35.91.* for a network of computers at his house. Larger organizations can get a "Class B" network where the first two bytes identify the network and the last two bytes identify each of up to 64 thousand individual workstations. Yale's Class B network is 130.132, so all computers with IP address 130.132.*.* are connected through Yale.

The organization then connects to the Internet through one of a dozen regional or specialized network suppliers. The network vendor is given the subscriber network number and adds it to the routing configuration in its own machines and those of the other major network suppliers.

There is no mathematical formula that translates the numbers 192.35.91 or 130.132 into "Yale University" or "New Haven, CT." The machines that manage large regional networks or the central Internet routers managed by the National Science Foundation can only locate these networks by looking each network number up in a table. There are potentially thousands of Class B networks, and millions of Class C networks, but computer memory costs are low, so the tables are reasonable. Customers that connect to the Internet, even customers as large as IBM, do not need to maintain any information on other networks. They send all external data to the regional carrier to which they subscribe, and the regional carrier maintains the tables and does the appropriate routing.

New Haven is in a border state; split 50-50 between the Yankees and the Red Sox. In this spirit, Yale recently switched its connection from the Middle Atlantic regional network to the New England carrier. When the switch occurred, tables in the other regional areas and in the national spine had to be updated, so that traffic for 130.132 was routed through Boston instead of New Jersey. The large network carriers handle the paperwork and can perform such a switch given sufficient notice. During a conversion period, the university was connected to both networks so that messages could arrive through either path.

2.2.2 Subnets

Although the individual subscribers do not need to tabulate network numbers or provide explicit routing, it is convenient for most Class B networks to be internally managed as a much smaller and simpler version of the larger network organizations. It is common to subdivide the two bytes available for internal assignment into a one byte department number and a one byte workstation ID.

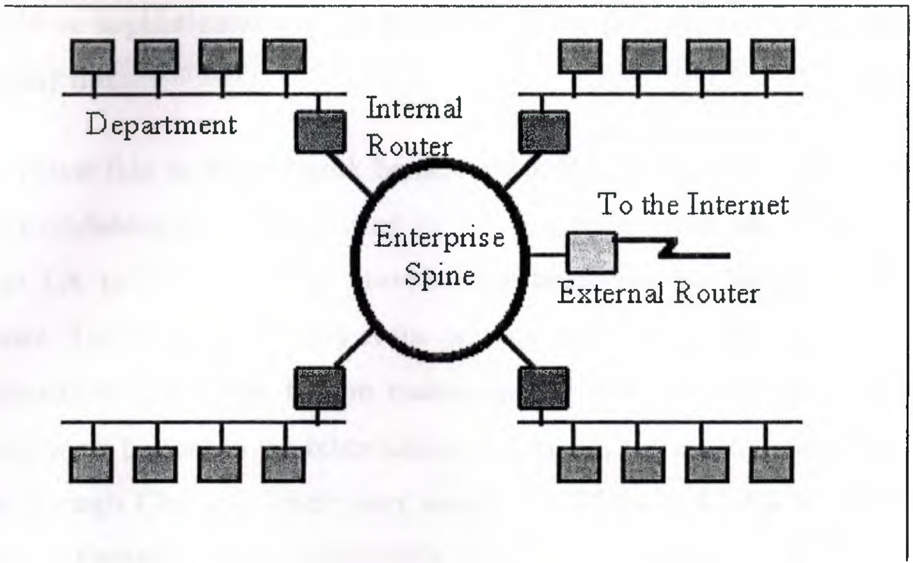


Fig no. 2.2

The enterprise network is built using commercially available TCP/IP router boxes. Each router has small tables with 255 entries to translate the one byte department number into selection of a destination Ethernet connected to one of the routers. Messages to the PC Lube and Tune server (130.132.59.234) are sent through the national and New England regional networks based on the 130.132 part of the number. Arriving at Yale, the 59 department ID selects an Ethernet connector in the C& IS building. The 234 selects a particular workstation on that LAN. The Yale network must be updated as new Ethernets and departments are added, but it is not affected by changes outside the university or the movement of machines within the department.

2.2.3 A Uncertain Path

Every time a message arrives at an IP router, it makes an individual decision about where to send it next. There is concept of a session with a reselected path for all traffic. Consider a company with facilities in New York, Los Angeles, Chicago and Atlanta. It could build a network from four phone lines forming a loop (NY to Chicago to LA to Atlanta to NY). A message arriving at the NY router could go to LA via either Chicago or Atlanta. The reply could come back the other way.

How does the router make a decision between routes? There is no correct answer. Traffic could be routed by the "clockwise" algorithm (go NY to Atlanta, LA to Chicago). The routers could alternate, sending one message to Atlanta and the next to

Chicago. More sophisticated routing measures traffic patterns and sends data through the least busy link.

If one phone line in this network breaks down, traffic can still reach its destination through a roundabout path. After losing the NY to Chicago line, data can be sent NY to Atlanta to LA to Chicago. This provides continued service though with degraded performance. This kind of recovery is the primary design feature of IP. The loss of the line is immediately detected by the routers in NY and Chicago, but somehow this information must be sent to the other nodes. Otherwise, LA could continue to send NY messages through Chicago, where they arrive at a "dead end." Each network adopts some Router Protocol which periodically updates the routing tables throughout the network with information about changes in route status.

If the size of the network grows, then the complexity of the routing updates will increase as will the cost of transmitting them. Building a single network that covers the entire US would be unreasonably complicated. Fortunately, the Internet is designed as a Network of Networks. This means that loops and redundancy are built into each regional carrier. The regional network handles its own problems and reroutes messages internally. Its Router Protocol updates the tables in its own routers, but no routing updates need to propagate from a regional carrier to the NSF spine or to the other regions (unless, of course, a subscriber switches permanently from one region to another).

2.2.4 Undiagnosed Problems

IBM designs its SNA networks to be centrally managed. If any error occurs, it is reported to the network authorities. By design, any error is a problem that should be corrected or repaired. IP networks, however, were designed to be robust. In battlefield conditions, the loss of a node or line is a normal circumstance. Casualties can be sorted out later on, but the network must stay up. So IP networks are robust. They automatically (and silently) reconfigure themselves when something goes wrong. If there is enough redundancy built into the system, then communication is maintained.

In 1975 when SNA was designed, such redundancy would be prohibitively expensive, or it might have been argued that only the Defense Department could afford it. Today, however, simple routers cost no more than a PC. However, the TCP/IP design that, "Errors are normal and can be largely ignored," produces problems of its own.

Data traffic is frequently organized around "hubs," much like airline traffic. One could imagine an IP router in Atlanta routing messages for smaller cities throughout the Southeast. The problem is that data arrives without a reservation. Airline companies experience the problem around major events, like the Super Bowl. Just before the game, everyone wants to fly into the city. After the game, everyone wants to fly out. Imbalance occurs on the network when something new gets advertised. Adam Curry announced the server at "mtv.com" and his regional carrier was swamped with traffic the next day. The problem is that messages come in from the entire world over high speed lines, but they go out to mtv.com over what was then a slow speed phone line.

Occasionally a snow storm cancels flights and airports fill up with stranded passengers. Many go off to hotels in town. When data arrives at a congested router, there is no place to send the overflow. Excess packets are simply discarded. It becomes the responsibility of the sender to retry the data a few seconds later and to persist until it finally gets through. This recovery is provided by the TCP component of the Internet protocol.

TCP was designed to recover from node or line failures where the network propagates routing table changes to all router nodes. Since the update takes some time, TCP is slow to initiate recovery. The TCP algorithms are not tuned to optimally handle packet loss due to traffic congestion. Instead, the traditional Internet response to traffic problems has been to increase the speed of lines and equipment in order to say ahead of growth in demand.

TCP treats the data as a stream of bytes. It logically assigns a sequence number to each byte. The TCP packet has a header that says, in effect, "This packet starts with byte 379642 and contains 200 bytes of data." The receiver can detect missing or incorrectly sequenced packets. TCP acknowledges data that has been received and retransmits data that has been lost. The TCP design means that error recovery is done end-to-end between the Client and Server machine. There is no formal standard for tracking problems in the middle of the network, though each network has adopted some ad hoc tools.

2.2.5 Levels

There are three levels of TCP/IP knowledge. Those who administer a regional or national network must design a system of long distance phone lines, dedicated routing devices, and very large configuration files. They must know the IP numbers and physical locations of thousands of subscriber networks. They must also have a formal network monitor strategy to detect problems and respond quickly.

Each large company or university that subscribes to the Internet must have an intermediate level of network organization and expertise. Half dozen routers might be configured to connect several dozen departmental LANs in several buildings. All traffic outside the organization would typically be routed to a single connection to a regional network provider.

However, the end user can install TCP/IP on a personal computer without any knowledge of either the corporate or regional network. Three pieces of information are required:

1. The IP address assigned to this personal computer
2. The part of the IP address (the subnet mask) that distinguishes other machines on the same LAN (messages can be sent to them directly) from machines in other departments or elsewhere in the world (which are sent to a router machine)
3. The IP address of the router machine that connects this LAN to the rest of the world.

In the case of the PCLT server, the IP address is 130.132.59.234. Since the first three bytes designate this department, a "subnet mask" is defined as 255.255.255.0 (255 is the largest byte value and represents the number with all bits turned on). It is a Yale convention (which we recommend to everyone) that the router for each department have station number 1 within the department network. Thus the PCLT router is 130.132.59.1. Thus the PCLT server is configured with the values:

- My IP address: 130.132.59.234
- Subnet mask: 255.255.255.0
- Default router: 130.132.59.1

The subnet mask tells the server that any other machine with an IP address beginning 130.132.59.* is on the same department LAN, so messages are sent to it directly. Any IP address beginning with a different value is accessed indirectly by sending the message through the router at 130.132.59.1 (which is on the departmental LAN).

2.3 X.25 Background

X.25 network devices fall into three general categories: data terminal equipment (DTE), data circuit-terminating equipment (DCE), and packet-switching exchange (PSE). Data terminal equipment devices are end systems that communicate across the X.25 network. They are usually terminals, personal computers, or network hosts, and are located on the premises of individual subscribers. DCE devices are communications devices, such as modems and packet switches that provide the interface between DTE devices and a PSE, and are generally located in the carrier's facilities. PSEs are switches that compose the bulk of the carrier's network. They transfer data from one DTE device to another through the X.25 PSN. Figure illustrates the relationships among the three types of X.25 network devices.

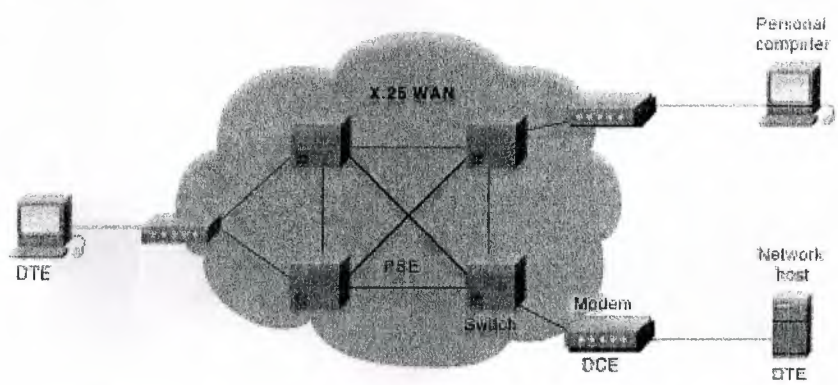


Figure 2.3 DTEs, DCEs, and PSEs Make Up an X.25 Network

2.3.1 X.25 Session Establishment

X.25 sessions are established when one DTE device contacts another to request a communication session. The DTE device that receives the request can either accept or refuse the connection. If the request is accepted, the two systems begin full-duplex information transfer. Either DTE device can terminate the connection. After the session is terminated, any further communication requires the establishment of a new session.

2.3.2 X.25 Virtual Circuits

A virtual circuit is a logical connection created to ensure reliable communication between two network devices. A virtual circuit denotes the existence of a logical, bidirectional path from one DTE device to another across an X.25 network. Physically, the connection can pass through any number of intermediate nodes, such as DCE devices and PSEs. Multiple virtual circuits (logical connections) can be multiplexed onto a single physical circuit (a physical connection). Virtual circuits are demultiplexed at the remote end, and data is sent to the appropriate destinations. Illustrates four separate virtual circuits being multiplexed onto a single physical circuit.

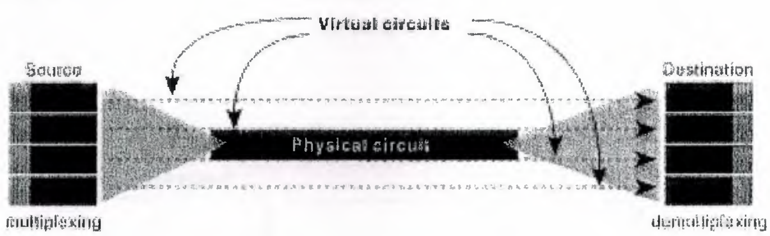


Figure 2.4 Virtual Circuits Can Be Multiplexed onto a Single Physical Circuit

Two types of X.25 virtual circuits exist: switched and permanent. Switched virtual circuits (SVCs) are temporary connections used for sporadic data transfers. They require that two DTE devices establish, maintain, and terminate a session each time the devices need to communicate. Permanent virtual circuits (PVCs) are permanently established connections used for frequent and consistent data transfers. PVCs do not require that sessions be established and terminated. Therefore, DTEs can begin transferring data whenever necessary because the session is always active.

The basic operation of an X.25 virtual circuit begins when the source DTE device specifies the virtual circuit to be used (in the packet headers) and then sends the packets to a locally connected DCE device. At this point, the local DCE device examines the packet headers to determine which virtual circuit to use and then sends the packets to the closest PSE in the path of that virtual circuit. PSEs (switches) pass the traffic to the next intermediate node in the path, which may be another switch or the remote DCE device.

When the traffic arrives at the remote DCE device, the packet headers are examined and the destination address is determined. The packets are then sent to the destination DTE device. If communication occurs over an SVC and neither device has additional data to transfer, the virtual circuit is terminated.

2.3.3 The X.25 Protocol Suite

The X.25 protocol suite maps to the lowest three layers of the OSI reference model. The following protocols are typically used in X.25 implementations: Packet-Layer Protocol (PLP), Link Access Procedure, Balanced (LAPB), and those among other physical-layer serial interfaces (such as EIA/TIA-232, EIA/TIA-449, EIA-530, and G.703). Maps the key X.25 protocols to the layers of the OSI reference model.

Reference Model

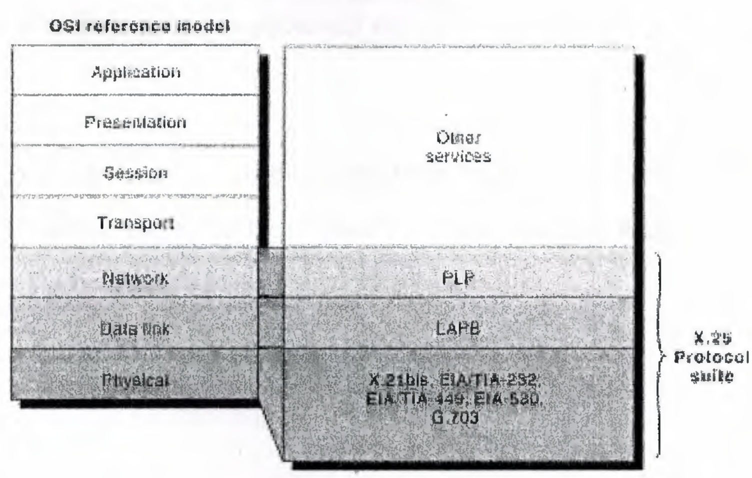


Figure 2.5 Key X.25 Protocols Map to the Three Lower Layers of the OSI

2.3.4 Packet-Layer Protocol

PLP is the X.25 network layer protocol. PLP manages packet exchanges between DTE devices across virtual circuits. PLPs also can run over Logical Link Control 2 (LLC2) implementations on LANs and over Integrated Services Digital Network (ISDN) interfaces running Link Access Procedure on the D channel (LAPD).

The PLP operates in five distinct modes: call setup, data transfer, idle, call clearing, and restarting.

Call setup mode is used to establish SVCs between DTE devices. A PLP uses the X.121 addressing scheme to set up the virtual circuit. The call setup mode is executed on a per-virtual-circuit basis, which means that one virtual circuit can be in call setup mode while another is in data transfer mode. This mode is used only with SVCs, not with PVCs.

Data transfer mode is used for transferring data between two DTE devices across a virtual circuit. In this mode, PLP handles segmentation and reassembly, bit padding, and error and flow control. This mode is executed on a per-virtual-circuit basis and is used with both PVCs and SVCs.

Idle mode is used when a virtual circuit is established but data transfer is not occurring.
It is executed on a per-virtual-circuit basis and is used only with SVCs.

Call clearing mode is used to end communication sessions between DTE devices and to terminate SVCs. This mode is executed on a per-virtual-circuit basis and is used only with SVCs.

Restarting mode is used to synchronize transmission between a DTE device and a locally connected DCE device. This mode is not executed on a per-virtual-circuit basis. It affects all the DTE device's established virtual circuits.

Four types of PLP packet fields exist:

- General Format Identifier (GFI)—Identifies packet parameters, such as whether the packet carries user data or control information, what kind of windowing is being used, and whether delivery confirmation is required.
- Logical Channel Identifier (LCI)—identifies the virtual circuit across the local DTE/DCE interface.
- Packet Type Identifier (PTI)—identifies the packet as one of 17 different PLP packet types.
- User Data—Contains encapsulated upper-layer information. This field is present only in data packets. Otherwise, additional fields containing control information are added.

2.4 Back ground of DVB-T (Channel coding and modulation)

The outer coding and interleaving shall be performed on the input packet structure. Reed-Solomon RS (204,188, t = 8) shortened code derived from the original systematic RS (255,239, t = 8) code, shall be applied to each randomized transport packet (188 byte) of to generate an error protected packet. Reed-Solomon coding shall also be applied to the packet sync byte, either non-inverted (i.e. 47HEX) or inverted (i.e. B8HEX).

Code Generator Polynomial

G(x) = (x+λ^0) (x+λ^1).....(x+λ^15)

Where λ=02 Hex

Field Generator Polynomial

P(x) = x^8+x^4+x^3+x^2+1

The shortened Reed-Solomon code may be implemented by adding 51 bytes, all set to zero, before the information bytes at the input of an RS (255,239, t = 8) encoder. After the RS coding procedure these null bytes shall be discarded, leading to a RS code word of N = 204 bytes. Following the conceptual scheme of figure, convolution byte-wise interleaving with depth I = 12 shall be applied to the error protected packets (see figure 2.6. These results in the interleaved data structure (see figure 2.9). The interleaved data bytes shall be composed of error protected packets and shall be delimited by inverted or non-inverted MPEG-2 sync bytes (preserving the periodicity of 204 bytes). The interleaves may be composed of I = 12 branches, cyclically connected to the input byte-stream by the input switch. Each branch j shall be a First-In, First-Out (FIFO) shift register, with depth j × M cells where M = 17 = N/I, N = 204. The cells of the FIFO shall contain 1 byte, and the input and output switches shall be synchronized. For synchronization purposes, the SYNC bytes and the SYNC bytes shall always be routed in the branch "0" of the interleaves (corresponding to a null delay).

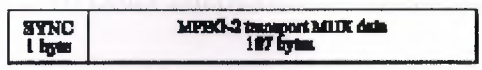


Figure 2.6 MPEG-2 Transport MUX Packets

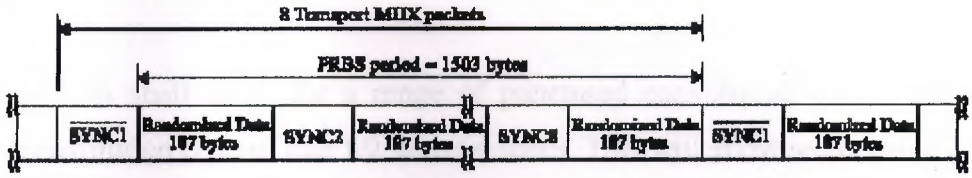


Figure 2.7 Randomized Transport packets

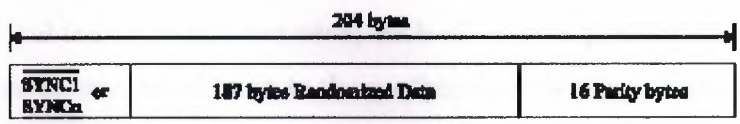


Figure 2.8 Reed Solomon RS (204,188) error packeted packet

Figure 2.9 Data structure after interleaving

SYNC1 is the non randomized complemented sync byte and Sync is the non randomized sync byte, n = 2, 3, 8.

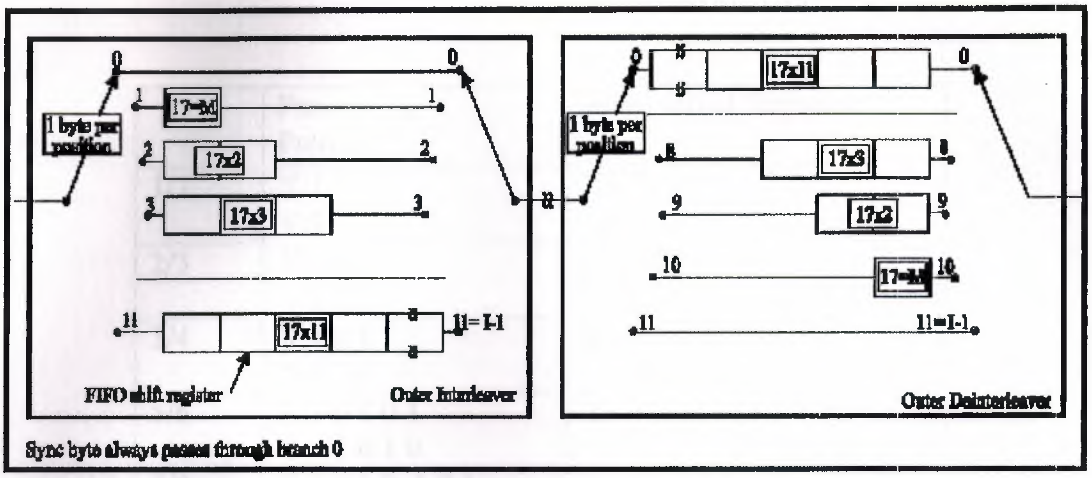


Figure 2.10 Outer interleaver and Deinterleaver

2.4.1 Inner coding

The system shall allow for a range of punctured convolution codes, based on a mother convolution code of rate 1/2 with 64 states. This will allow selection of the most appropriate level of error correction for a given service or data rate in either non-hierarchical or hierarchical transmission mode. The generator polynomials of the mother code are G1 = 171OCT for X output and G2 = 133OCT for Y output. If two level hierarchical transmissions are used; each of the two parallel channel encoders can have its own code rate. In addition to the mother code of rate 1/2 the system shall allow punctured rates of 2/3, 3/4, 5/6 and 7/8. The punctured convolution code shall be used as given in table 2.5. See also figure 2.5. In this table X and Y refer to the

Two outputs of the convolution encoder.

	Puncturing Pattern	Transmitted Sequence
1/2	X:1 Y:1	X1 Y1
2/3	X:1 0 Y:1 1	X1 Y1 Y2
3/4	X:1 0 1 Y:1 1 0	X1 Y1 Y2 X3
5/6	X:1 0 1 0 1 Y:1 1 0 1 0	X1 Y1 Y2 Y3 Y4 X5
7/8	X:1 0 0 0 1 0 1 Y:1 1 1 1 0 1 0	X1 Y1 Y2 Y3 Y4 Y5 Y6 X7

Table 2.5

X1 is sent first. At the start of a super-frame the MSB of SYNC or SYNC shall lie at the point labeled "data input" in figure 2.12. The first convolution ally encoded bit of a symbol always corresponds to X1.

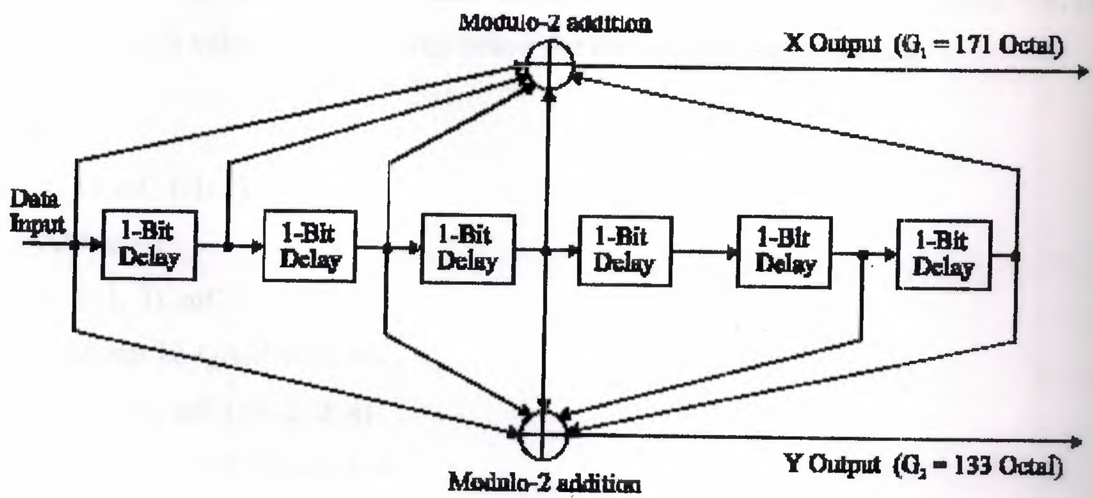


Figure 2.11 the mother Convolution Code

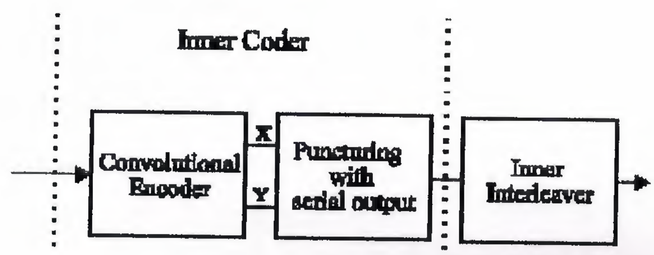


Figure 2.12 Inner coding and Interleaving

2.4.2 Inner interleaving

The inner interleaving consists of bit-wise interleaving followed by symbol interleaving. Both the bit-wise interleaving and the symbol interleaving processes are block-based.

2.4.3 Signal constellations and mapping

The system uses Orthogonal Frequency Division Multiplex (OFDM) transmission. All data carriers in one OFDM frame are modulated using QPSK, 16-QAM, 64-QAM, non-uniform 16-QAM or non-u uniform 64-QAM constellations. The constellations and the details of the Gray mapping applied to them. The exact proportions of the constellations depend on a parameter α, which can take the three values 1, 2 or 4. Minimum distance separating two constellation points carrying different HP-bit values divided by the minimum distance separating any two constellation points. Non-hierarchical transmission uses the same uniform constellation as the case with $\alpha = 4$, i.e. figure 2.13 with values of n, m given below for the various constellations.

QPSK

n€ (-1, 1), m€ (-1, 1)

16-QAM=α =1

n€ (-3,-1, 1, 3), m€ (-3,-1, 1, 3)

Non-Uniform 16-QAM with α=2

no (-4,-2, 2, 4), m€ (-4,-2, 2, 4)

Non-Uniform 16-QAM with α=4

n€ (-6,-4, 4, 6), m€ (-6,-4, 4, 6)

64QAM with α=1

no(-7,-5,-3,-1,1,3,5,7),m€(-7,-5,-3,-1,1,3,5,7)

Non-Uniform 64-QAM with α=2

n€(-8,-6,-4,-2,2,4,6,8),m€(-8,-6,-4,-2,2,4,6,8)

Non-Uniform 64-QAM with α =4

n€(-10,-8,-6,-4,4,6,8,10),m€(-10,-8,-6,-4,4,6,8,10)

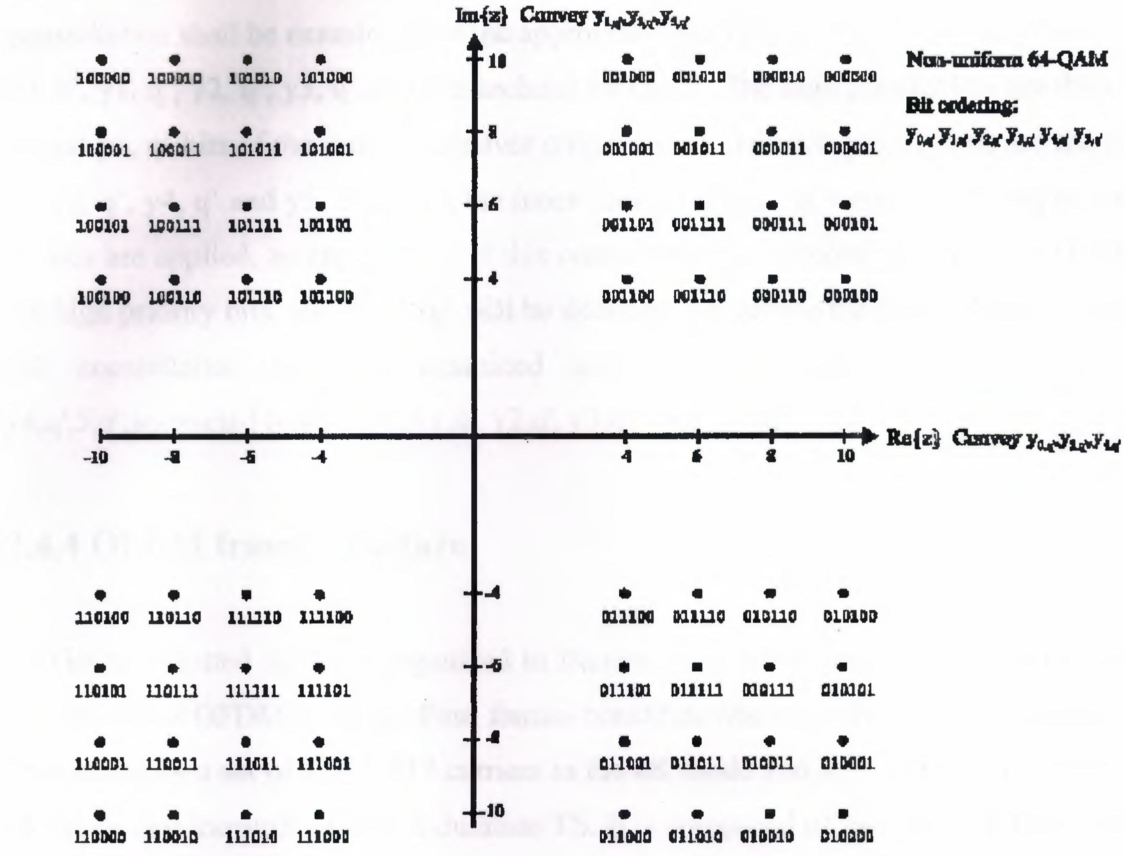


Figure 2.13 16-QAM and 64-QAM mapping with a=4

The you denote the bits representing a complex modulation symbol z. Non-hierarchical transmission: The data stream at the output of the inner interleaves consists of v bit words. These are mapped onto a complex number z, according to figure. Hierarchical transmission: In the case of hierarchical transmission, the data streams are formatted.

For hierarchical 16-QAM: The high priority bits are the y0, q' and y1, q' bits of the inner interleaver output words. The low priority bits are the y2, q' and y3, q' bits of the inner interleaver output words.

For example, the top left constellation point, corresponding to 1 000 represents y0, q' = 1, y1, q' = y2, q' = y3, q' = 0. If this constellation is decoded as if it were QPSK, the high priority bits, y0, q', y1, q' will be deduced. To decode the low priority bits, the full constellation shall be examined and the appropriate bits (y2, q', y3, q') extracted from y0, q', y1, q', y2, q', y3, q'. For hierarchical 64-QAM: The high priority bits are the y0, q' and y1, q' bits of the inner interleaver output words. The low priority bits are the y2, q', y3, q', y4, q' and y5, q' bits of the inner interleaver output words. The mappings of figures are applied, as appropriate. If this constellation is decoded as if it were QPSK, the high priority bits, y0, q', y1, q' will be deduced. To decode the low priority bits, the full constellation shall be examined and the appropriate bits (y2,q', y3,q', y4,q',5,q',)extracted from y0,q', y1,q', y2,q', y3,q', y4,q', y5,q'.

2.4.4 OFDM frame structure

The transmitted signal is organized in frames. Each frame has duration of TF, and consists of 68 OFDM symbols. Four frames constitute one super-frame. Each symbol is constituted by a set of K = 6 817 carriers in the 8K mode and K = 1 705 carriers in the 2K mode and transmitted with a duration TS. It is composed of two parts: a useful part with duration TU and a guard interval with duration. The guard interval consists in a cyclic continuation of the useful part, TU, and is inserted before it. The symbols in an OFDM frame are numbered from 0 to 67. All symbols contain data and reference information. Since the OFDM signal comprises many separately-modulated carriers, each symbol can in turn be considered to be divided into cells, each corresponding to the modulation carried on one carrier during one symbol. In addition to the transmitted data an OFDM frame contains:

- Scattered pilot cells;
- Continual pilot carriers;
- TPS carriers.

The pilots can be used for frame synchronization, frequency synchronization, time synchronization, channel estimation, transmission mode identification and can also be used to follow the phase noise. The carriers are indexed by k [Kmin; Kmax] and determined by Kmin = 0 andKmax = 1 704 in 2K mode and 6 816 in 8K mode respectively. The spacing between adjacent carriers is 1/TU while the spacing between carriers Kmin and Kmax are determined by (K-1)/TU. The numerical values for the

OFDM parameters for the 8K and 2K modes are given in tables 2.6 for 8 MHz channels. The values for the various time-related parameters are given in multiples of the elementary period T and in microseconds. The elementary period T is 7/64 μs for 8 MHz channels, 1/8 μs for 7 MHz channels and 7/48 μs for 6MHz channels.

Parameter	8k Mode	2k Mode
Value Of Carriers Number k	6817	1705
Value Of carriers Number k(min)	0	0
Value Of Carriers Number k(max)	6846	1704
Duration Tu	896μs	224μs
Carrier Spacing 1/Tu	1.116Hz	4.464Hz
Spacing Between carriers k(min) , K(max) (k-1)/Tu	7.61MHz	7.61MHz

Table 2.6

$$s(t) = \mathrm{Re}\left\{ e^{f2\pi fct} \sum_{0}^{\infty} \sum_{t-0}^{67} \sum_{k=k\min}^{k\max} c_{m,j,k} * \Psi_{m,j,k}(t) \right\}$$

$$where \Psi_{m,j,k}(t) = \left\{ e^{f2\pi k/Tu(t-\Delta-lxTs-68xmTs)} 1^{(l+68*m)xTs \le t \le (l+68*m+1)} \right\}$$

Where

k	denotes the carrier number
l	Denotes the OFDM symbol number
m	Denotes the transmission frame number
Ts	The symbol duration
Tu	The inverse of the carrier spacing
Δ	The duration of the guard interval
fc	The center frequency of the RF signal

c (m, j, k) Complex symbol for carrier k of the data symbol no.1 in frame number m

c (m, I, k) Complex symbol for carrier k of the data symbol no.2 in frame number m

c (m, 67, k) Complex symbol for carrier k of the data symbol no.68 in the frame m

2.4.4.1 Number of RS-packets per OFDM super-frame

The OFDM frame structure allows for an integer number of Reed-Solomon 204 byte packets to be transmitted in an OFDM super-frame, and therefore avoids the need for any stuffing, whatever the constellation, the guard interval length, the coding rate or the channel bandwidth may be. See table 2.8.

The first data byte transmitted in an OFDM super-frame shall be one of the SYNC/SYNC bytes.

Code rate	Qpsk 2k	Qpsk - 8k	16QAM 2k	16QAM - 8k	64 QAM 2k	64 QAM - 8k
1/2	252	1008	504	2016	756	3024
2/3	336	1344	672	2688	1008	4032
3/4	378	1512	756	3024	1134	4536
5/6	420	1680	840	3360	1260	5040
7/8	441	1764	882	3528	1323	5292

Table 2.8

2.4.5 Spectrum characteristics and spectrum mask

The OFDM symbols constitute a juxtaposition of equally-spaced orthogonal carriers. The amplitudes and phases of the data cell carriers are varying symbol by symbol according to the mapping process.

The power spectral density Pk (f) of each carrier at frequency:

$$fk = fc + k / Ts; [-(k - 1/2) \leq k(k - 1/2)]$$

Is defined by the following expression:

$$pk(f) = [\sin \pi(f - fk)Ts / \pi(f - fk)Ts]^{\wedge}2$$

The overall power spectral density of the modulated data cell carriers is the sum of the power spectral densities of all these carriers. A theoretical DVB transmission signal spectrum is illustrated in figure 2.14 (for 8 MHz channels). Because the OFDM symbol duration is larger than the inverse of the carrier spacing, the main lobe of the power spectral density of each carrier is narrower than twice the carrier spacing. Therefore the spectral density is not constant within the nominal bandwidth of *7,608 259* MHz for the 8K mode or 7,611 607MHz for the 2Kmode.

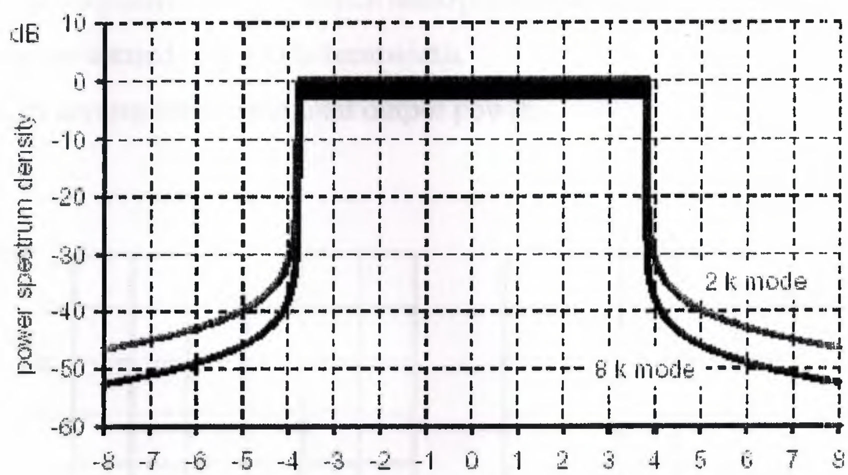


Figure 2.14 Frequency relative to center frequency fc

2.4.6 Out-of-band spectrum mask (for 8 MHz channels)

The level of the spectrum at frequencies outside the nominal bandwidth can be reduced by applying appropriate filtering. Spectrum masks for cases where a transmitter for digital terrestrial television is co-sited with and operating on a channel adjacent to, a transmitter for analogue television are given in figure 2.15 for the following analogue Television systems:

G / PAL / A2 and G / PAL / NICAM;

I / PAL /NICAM;

K / SECAM and K / PAL;

L / SECAM / NICAM.

The masks shown in figure 2.16 cover the minimum protection needed for analogue television where the analogue and the digital television transmitters are co-sited and are applicable for cases where:

- No polarization discrimination between digital and analogue television is used; and
- The radiated power from both transmitters is the same (analogue sync-peak power equal to total power from the digital television transmitter).

If the radiated powers from the two transmitters are not identical, proportional correction can be applied as follows:

Correction = minimum analogue erp - maximum digital erp.

Corrected breakpoints equal reference breakpoints plus correction (dB).

Power level measured in a 4 kHz bandwidth,

Where 0 dB corresponds to the total output power.

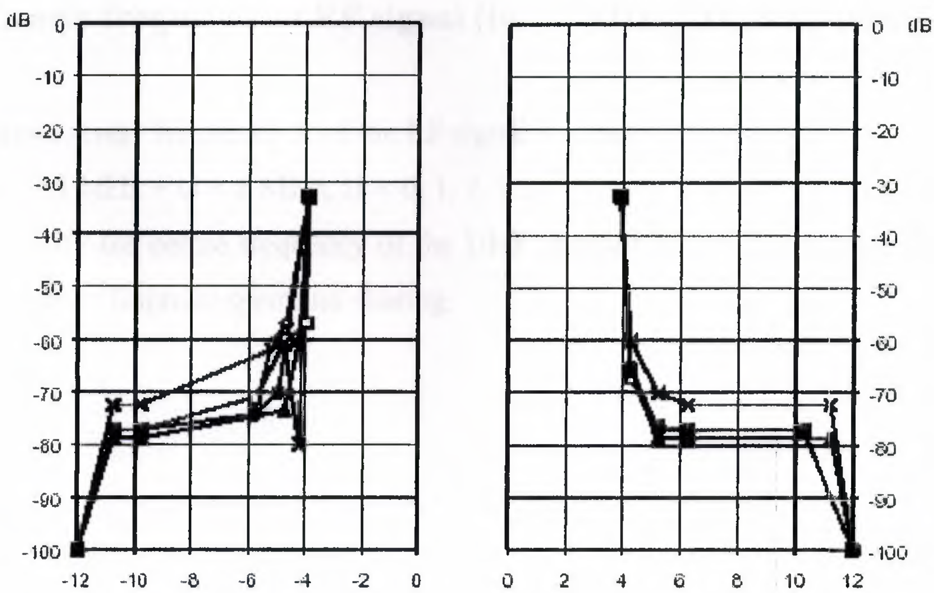


Figure 2.15 Frequency relative center of DVB-T channel

For critical cases such as television channels adjacent to other services (low power or receive only) a spectrum mask with higher of out-of-channel attenuation may be needed. A spectrum mask for critical cases is shown in figure 2.16.

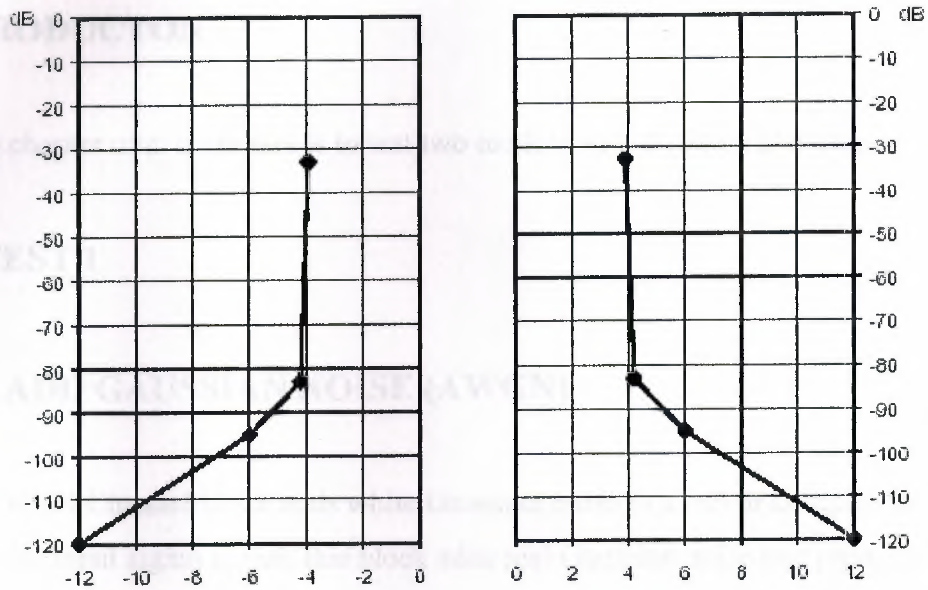


Figure 2.16 Spectrum mask for critical cases

2.4.7 Centre frequency of RF signal (for 8 MHz UHF channels)

The nominal centre frequency fc of the RF signal is given by:

470 MHz + 4 MHz + i1 × 8 MHz, i1 = 0, 1, 2, 3 ...

This is exactly the centre frequency of the UHF channel in use. This centre frequency may be offset to improve spectrum sharing.

Chapter 3

EXPERIMENTS

INTRODUCTON

In this chapter over main aim is to test two models with different channels .

3.1 TEST 1

3.1.1 ADD GAUSSIAN NOISE (AWGN)

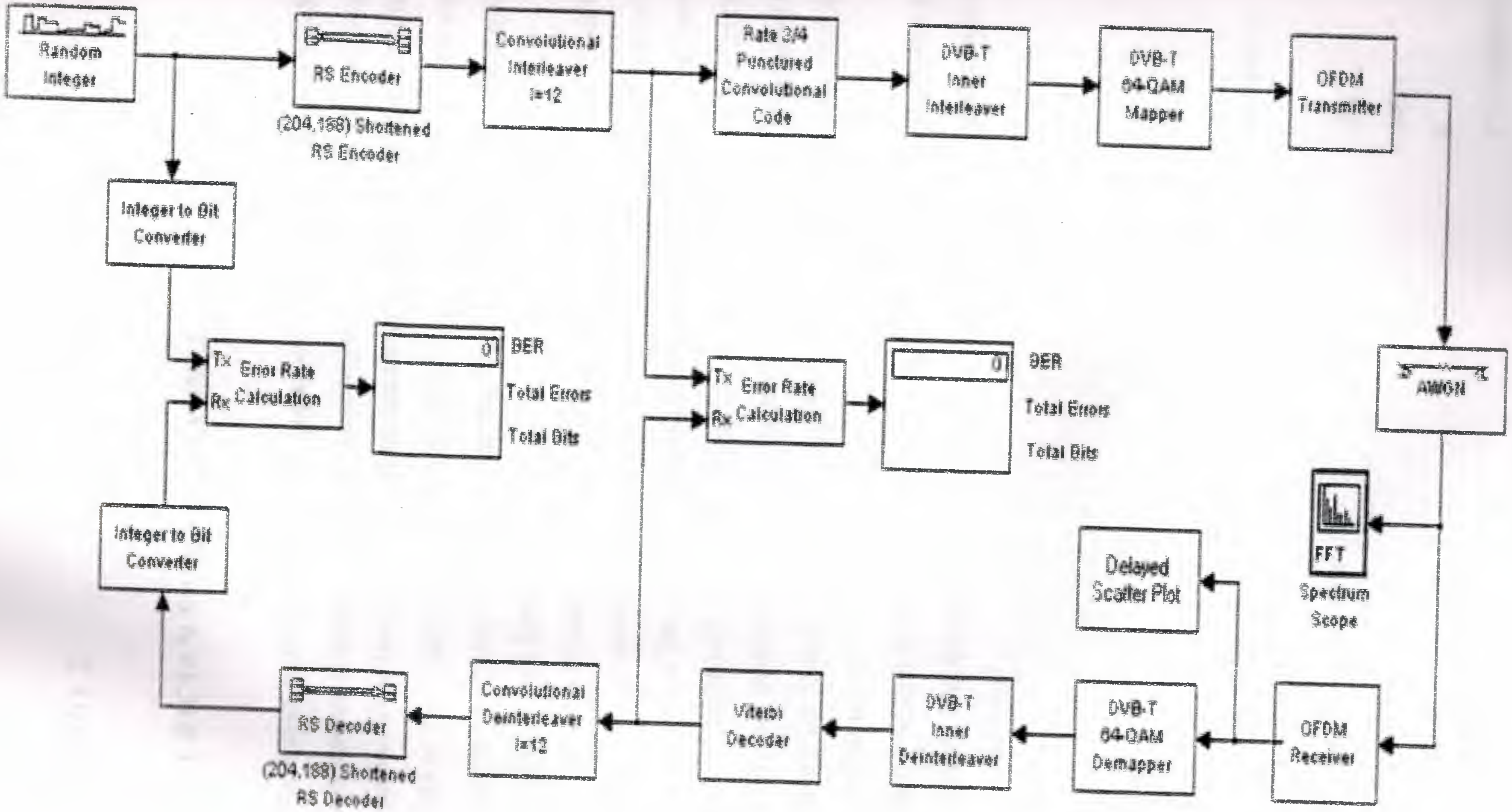
The AWGN Channel block adds white Gaussian noise to a real or complex input signal. When the input signal is real, this block adds real Gaussian noise and produces a real output signal. When the input signal is complex, this block adds complex Gaussian noise and produces a complex output signal. This block inherits its sample time from the input signal.

This block uses the DSP Block set's Random Source block to generate the noise. The Initial seed parameter in this block initializes the noise generator. Initial seed can be either a scalar or a vector whose length matches the number of channels in the input signal. For details on Initial seed, see the Random Source block reference page in the DSP Block set User's Guide.

DATA

- Sample time 1/3499081s
- Signal noise ratio (SNR) 15db
- Samples per frame 188
- M-aryl number 256
- Input signal power 1/2048W

2k Mode, Nonhierarchical Transmission



3.2 TEST 2

3.2.1 RICIAN FADING CHANNEL

The Rician Fading Channel block implements a baseband simulation of a Rician fading propagation channel. This block is useful for modeling mobile wireless communication systems when the transmitted signal can travel to the receiver along a dominant line-of-sight or direct path. If the signal can travel along a line-of-sight path and also along other fading paths, then you can use this block in parallel with the Multipath Rayleigh Fading Channel block.

The input can be either a scalar or a frame-based column vector. The input is a complex signal. Fading causes the signal to spread and become diffuse. The K-factor parameter, which is part of the statistical description of the Rician distribution, represents the ratio between direct-path (unspread) power and diffuse power. The ratio is expressed linearly, not in decibels. While the Gain parameter controls the overall gain through the channel, the K-factor parameter controls the gain's partition into direct and diffuse components.

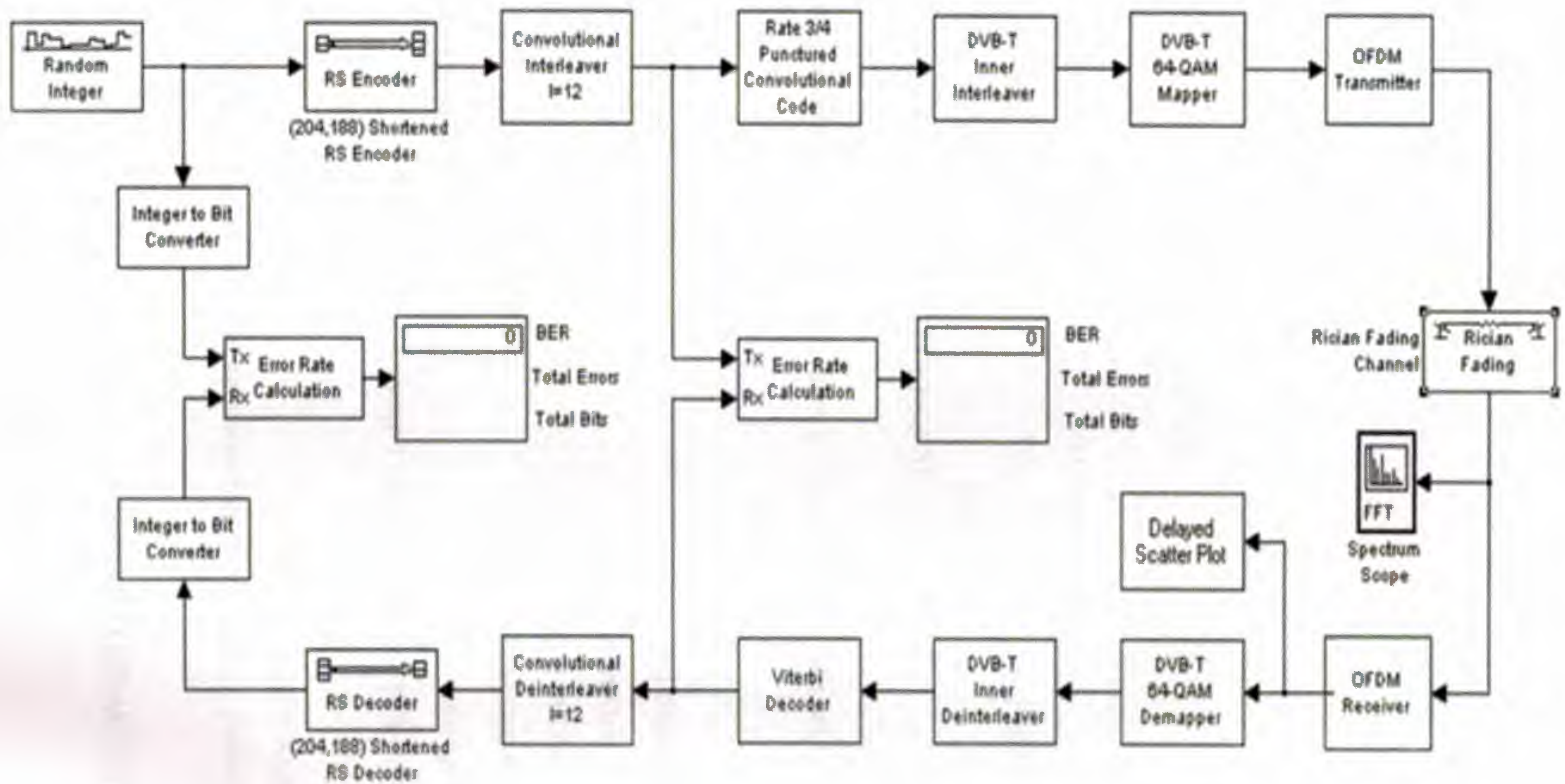
Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. The Jakes PSD (power spectral density) determines the spectrum of the Rician process.

DATA

• Sample time	1/3499081s
• K-Factor	1
• Samples per frame	188
• Doppler Shift	40Hz
• Gain	0
• Delay	3*1/3499081s

Digital Video Broadcasting-Terrestrial

2k Mode, Nonhierarchical Transmission



Chapter 4

Results

4.1 Result (AWGN) channel

In this channel we take the measurement three times for three different SNR values and analyze the results.

Test 1(a)

In this test the block parameters are as follows

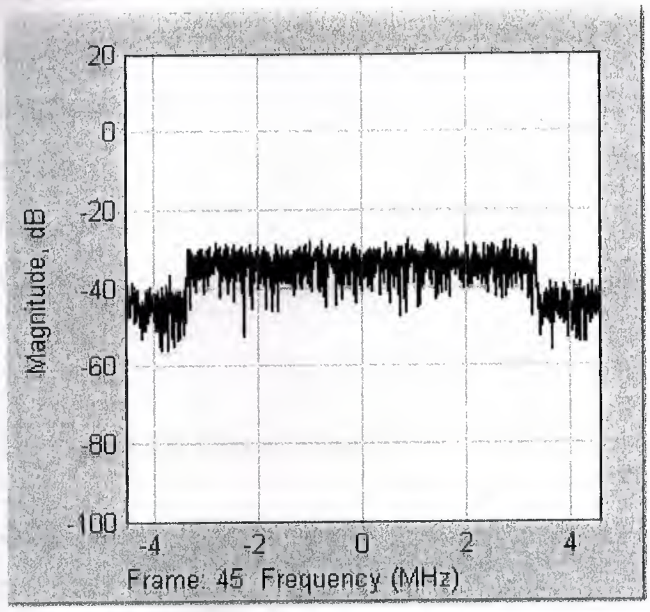
- Sample time 1/3499081s
- Signal noise ratio (SNR) 10dB
- Samples per frame 188
- M-ary number 256
- Input signal power 1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

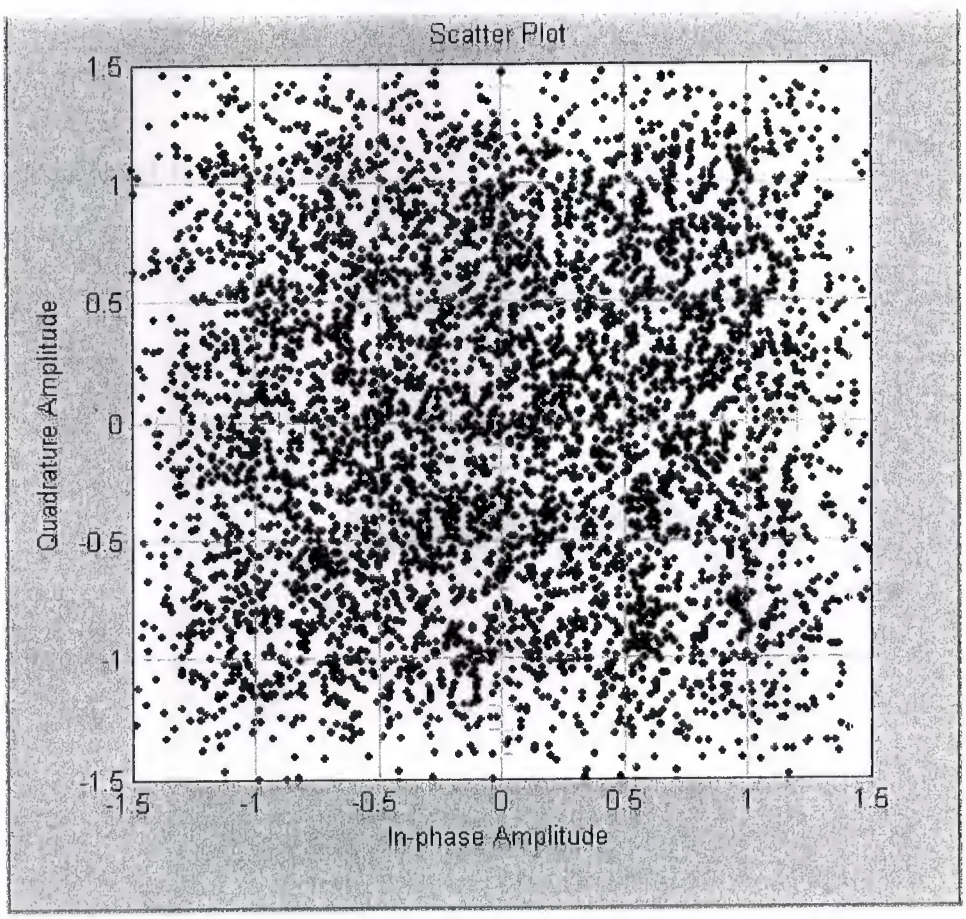
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.4699	0.4746
TOTAL ERRORS	1.38e+005	1,142e+005
TOTAL BITS	2.938e+005	2.542e+005

TABLE 1

Power Spectral Density Plot



Scatter Plot



TEST 1(b).

In this test over block parameters are as follows

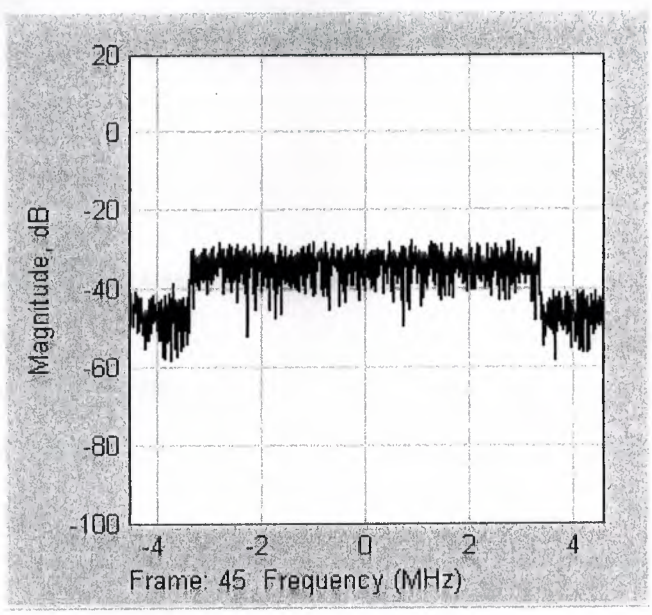
- Sample time 1/3499081s
- Signal noise ratio (SNR) 12dB
- Samples per frame 188
- M-ary number 256
- Input signal power 1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

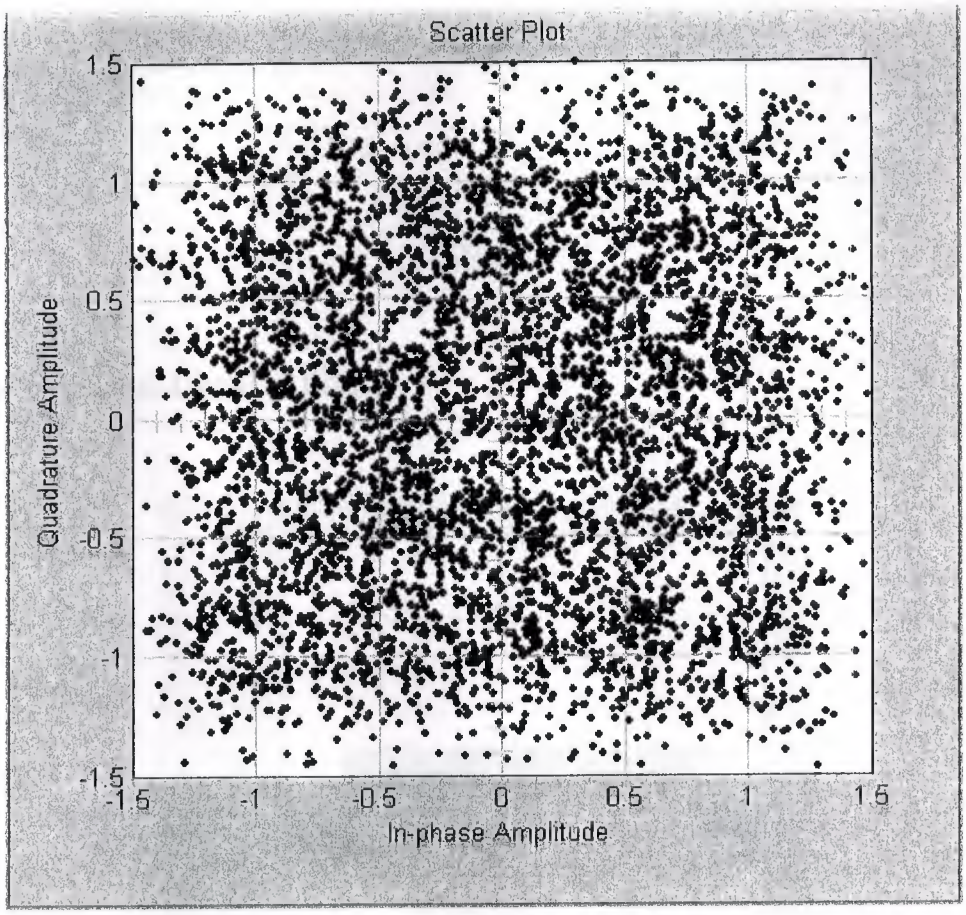
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.3886	0.3961
TOTAL ERRORS	1.142e+005	1.007e+005
TOTAL BITS	2.938e+005	2.542e+005

TABLE 2

Power Spectral Density Plot



Scatter Plot



TEST 1(c).

In this test over block parameters are as follows

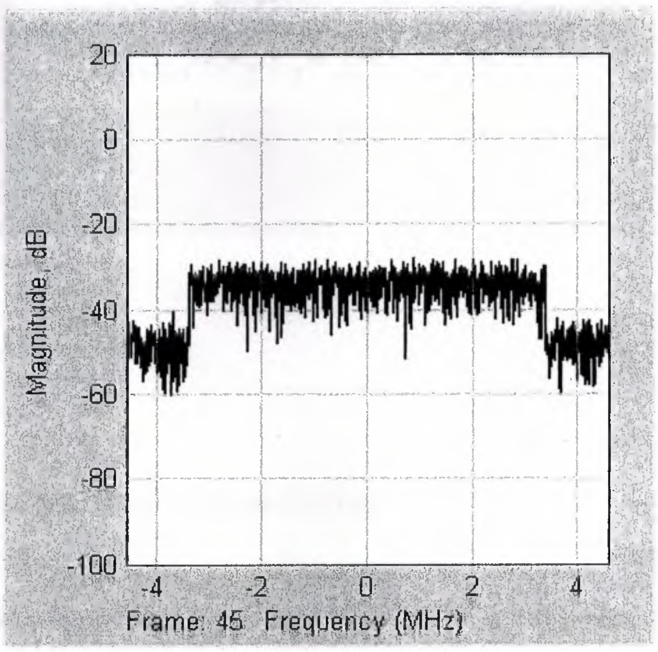
• Sample time	1/3499081s
• Signal noise ratio (SNR)	14dB
• Samples per frame	188
• M-aryl number	256
• Input signal power	1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

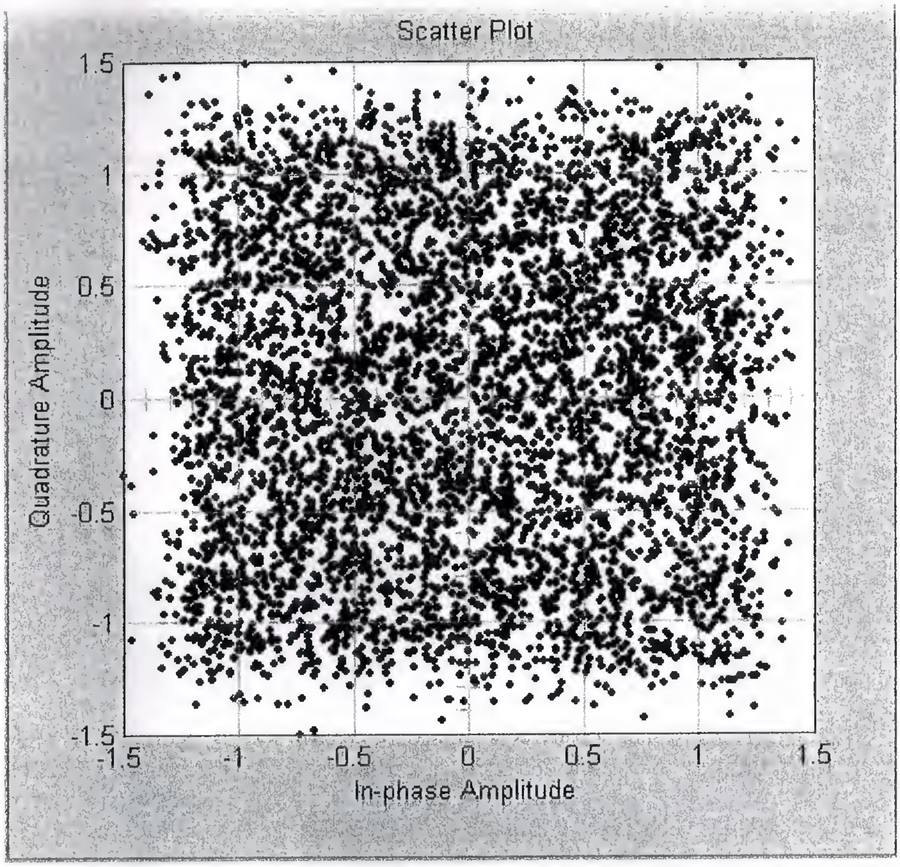
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.1642	0.1690
TOTAL ERRORS	4.825e+004	4.296e+004
TOTAL BITS	2.938e+005	2.542e+005

TABLE 3

Power Spectral Density Plot



Scatter Plot



TEST 1(d).

In this test over block parameters are as follows

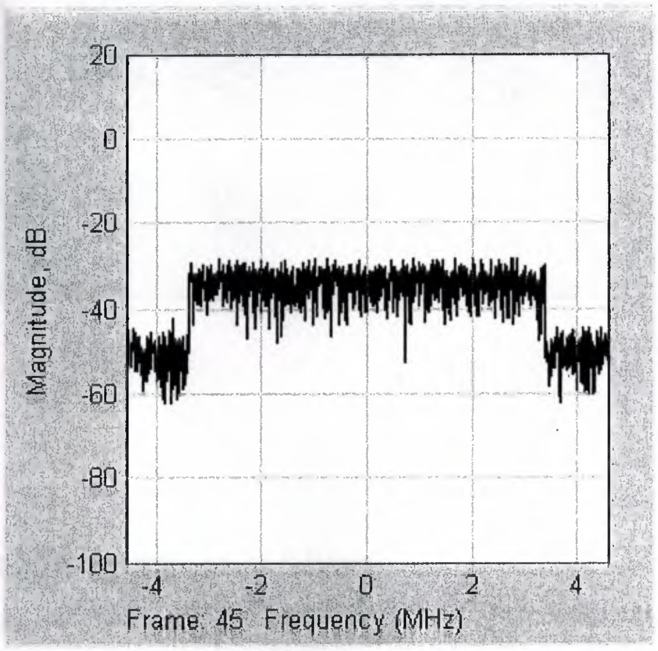
• Sample time	1/3499081s
• Signal noise ratio (SNR)	16dB
• Samples per frame	188
• M-aryl number	256
• Input signal power	1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

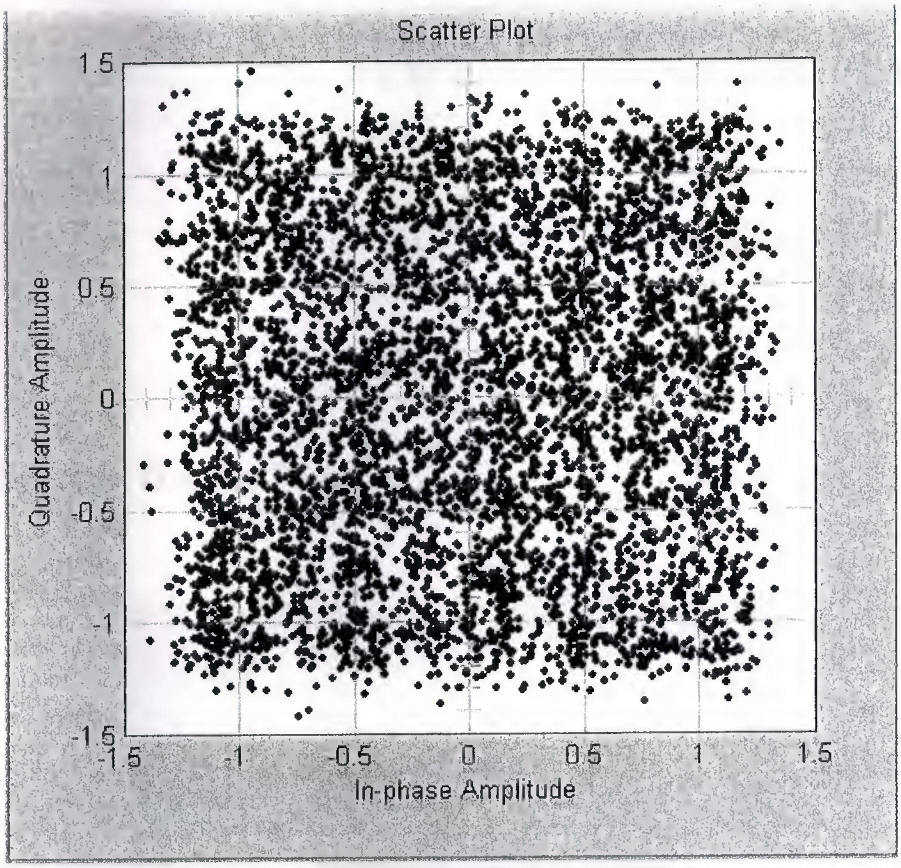
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.01179	0.002652
TOTAL ERRORS	3463	674
TOTAL BITS	2.938e+005	2.542e+005

TABLE 4

Power Spectral Density Plot



SCATTER PLOT



TEST 1(e).

In this test over block parameters are as follows

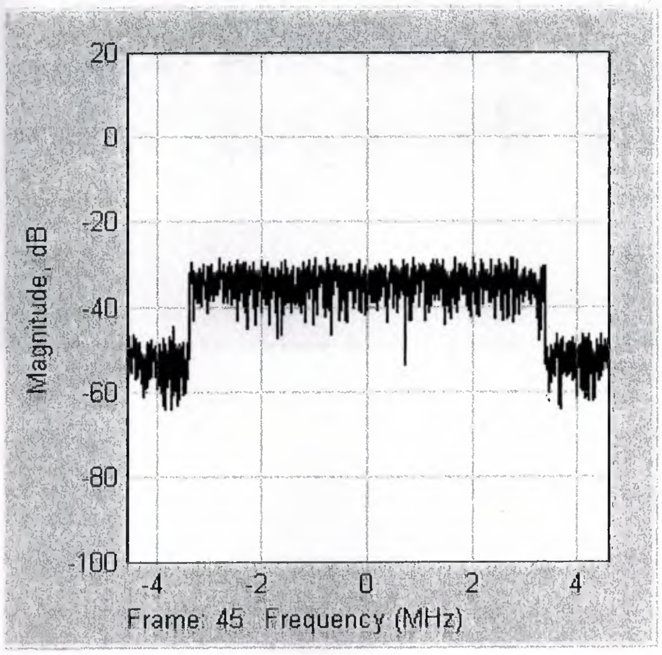
- Sample time 1/3499081s
- Signal noise ratio (SNR) 18dB
- Samples per frame 188
- M-aryl number 256
- Input signal power 1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

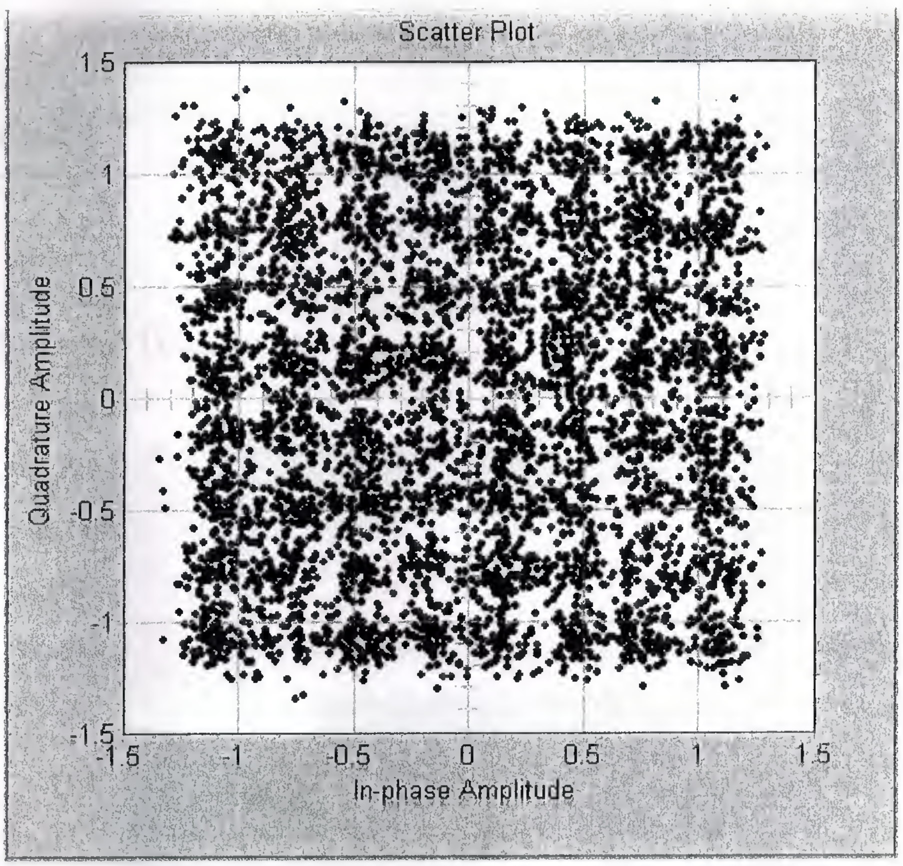
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.0001974	0
TOTAL ERRORS	58	0
TOTAL BITS	2.938e+005	2.542e+005

TABLE 5

Power Spectral Density Plot



Scatter Plot



TEST 1(e).

In this test over block parameters are as follows

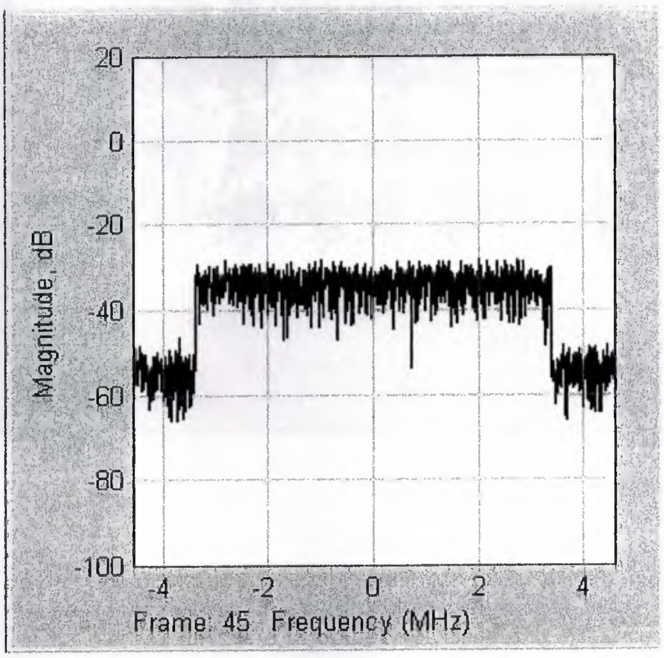
• Sample time	1/3499081s
• Signal noise ratio (SNR)	20dB
• Samples per frame	188
• M-aryl number	256
• Input signal power	1/2048W

We can show over result with the help of scatter plot, spectrum scope and table.

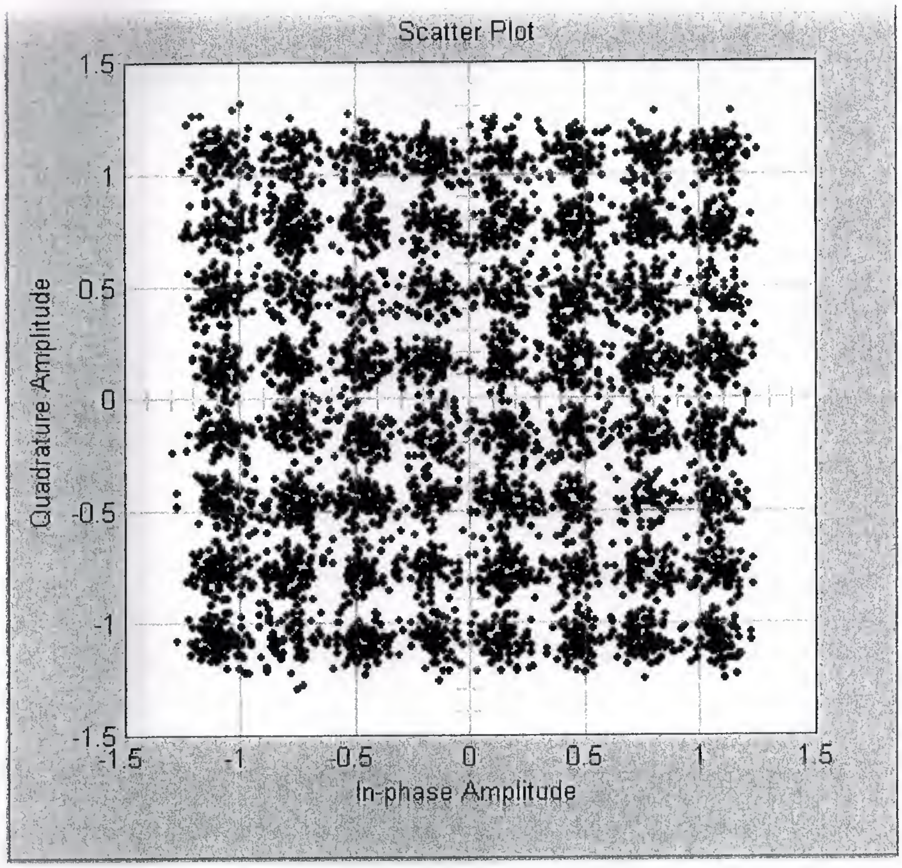
ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0	0
TOTAL ERRORS	0	0
TOTAL BITS	2.938e+005	2.542e+005

TABLE 6

Power Spectral Density Plot

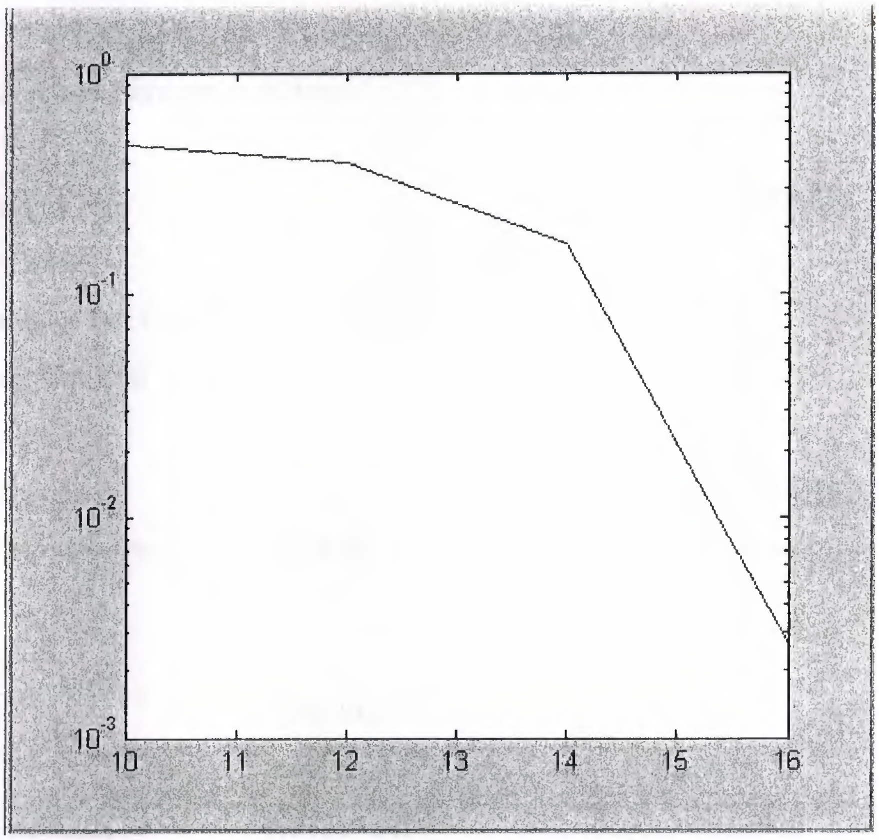


Scatter plot



We can compare the BER values with the help of a graph

Graph



The overall power spectral density of the modulated data cell carriers is the sum of the power spectral densities of all these carriers. Because the OFDM symbol duration is larger than the inverse of the carrier spacing, the main lobe of the power spectral density of each carrier is narrower than twice the carrier spacing. Therefore the spectral density is not constant within the nominal bandwidth of 7,608 259 MHz for the 8K mode or 7,611 607MHz for the 2Kmode .For more clear result we can see that in scatter plot we started from 10 db and finish it at 20db.We can see that as we increase the value of SNR value we are able to get better performance each time .In the end on 20 db we got the best result of over values and we can see each bit off information that has been transmitted.

4.2 Result Rican Fading

We test the channel three times by changing the parameters. In this channel we take over measurement by changing the Doppler value.

Test 2(a)

Over block parameters are as follows:

- Sample time 1/3499081s
- K-Factor 1
- Samples per frame 188
- Doppler Shift 20 Hz
- Gain 0
- Delay 3*1/3499081s

We can show over result with the help of a table, scatter plot and spectrum scope

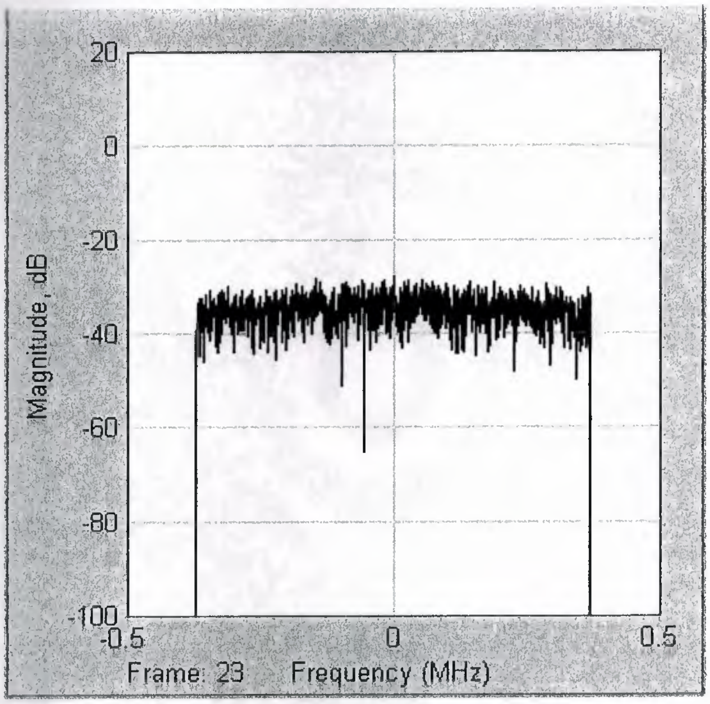
Table

ERRORS	BEFORE RS DECODER	AFTER RS DECODER
BER	0.4972	0.5002
TOTAL ERRORS	6.013e+005	5.491e+005
TOTAL BITS	1.209e+006	1.098e+006

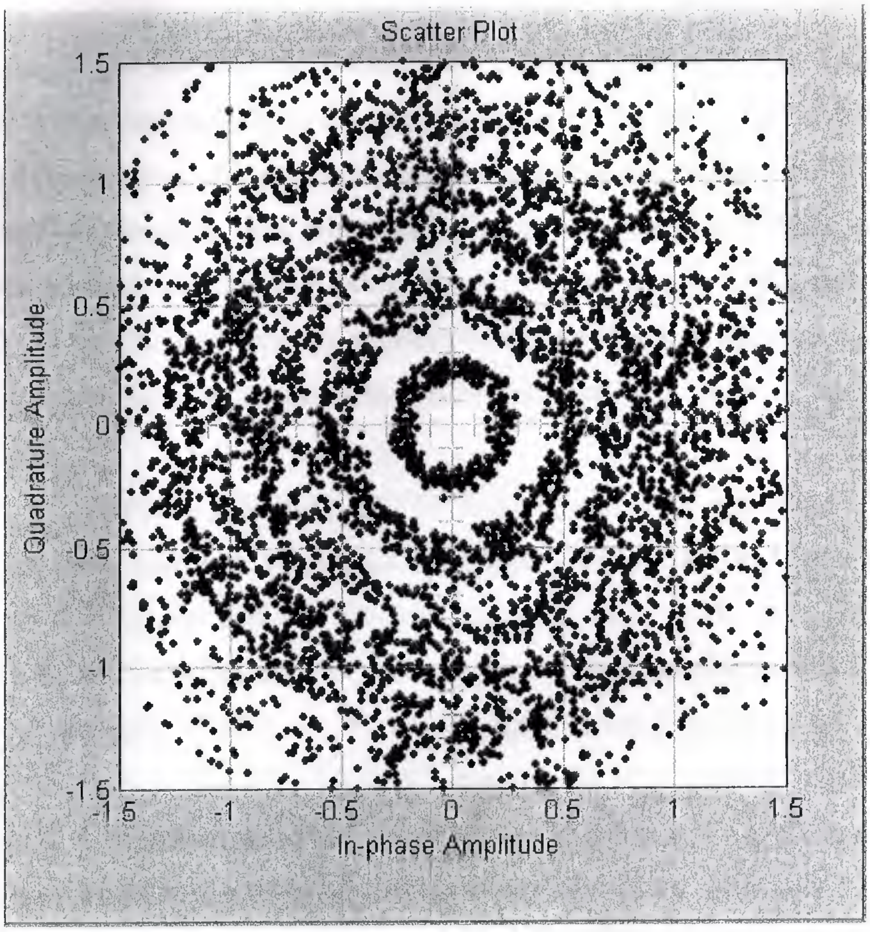
TABLE 7

Power Spectral Density Plot

NEAR EAST UNIVERSITY
LIBRARY
1988 - LEFKOŞA



Scatter plot



As from the result from RICIAN FADING channel we can see that there is no tolerance for DVB-T.This because of we are moving over TX an RX .To get a better tolerance then this the TX and the RX must not move and should be planted as a fix station.

CONCLUSION

The project of the DVB has resulted in a comprehensive list of technical and no technical documents describing solutions required by the university in order to be able to make the best use of the new technology of broadcasting digital signals.

Comparison between data protocols the project shows that ATM is simple and for its constant length of cells (53 bytes) helps us to send data easily and free of error .TCP/IP can also be used for over application but the length of the cells are not constant so it take more time to re-establish the information that cause us to wait for over transmission.

In over application (DVB-T) OFDM transmission with RS SOLOMON channel coding has been analyzed. The modulation techniques of 64-QAM have been used. In both cases RS-Encoder and RS- Decoder has been used. In the AWGN channel with SNR=10, we see that after the decoder the error bit does not change that much but after increasing the value of SNR to 20 we can see that error bit drops from 0.009876 to 0.But in the case of RICIAN FADING channel the total number of errors does not change a lot. This is because of the TX and RX is on move and we got no tolerance.

This means the performance of two considered channel test methods shows a big difference in results .The result shows that the AWGN has a better performance in DVB-T application.

FUTURE WORK

It was concluded that this software can be used by research students. And companies to develop and test new applications for DVB-T systems before going through the expensive prototyping process.

In future students can compare AWGN channel performance with the performance of Binary symmetric channel or Multipath rayleigh fading channel.

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[5] Fakhreddin Mamdov, Telecommunications (Lecture Note), Near East University Press, lefkosa, 1999

Abbreviations

RF	Radio Frequency
RS	Reed-Solomon
TV	Television
Lp	Low Priority Bit Stream
Hp	High Priority Bit Stream
IF	Intermediate Frequency
VHF	Very High Frequency
UHF	Ultra High Frequency
Tps	Transmission Parameter signaling
SFn	Signal Frequency Network
Hex	Hexadecimal Notation
FFt	Fast Fourier Transform
Msb	Most Significant Bit
Mux	Multiple x
DVB	Digital Video Broadcasting
Oct	Octal Notation
Dft	Discrete Fourier Transform
PAL	Phase Alternating Line
BER	Bit Error Ratio
Qam	Quadrature Amplitude Modulation
QEF	Quasi Error Free
ACI	Adjacent Channel Interference
QPSK	Quaternary Phase Shift Keying
OFDM	Orthogonal Frequency Division Multiplexing
DVB-T	Digital Video Broadcasting-Terrestrial
EDTV	Enhanced Definition Television
MPEG	Moving Picture Experts Group
HDTV	High Definition Television
LDTV	Limited Definition Television

IFFT	Inverse Fast Fourier transform
FIFO	First In First Out
SDTV	Standard Definition Television
SECAM	System Sequential Color A Memories
CBR	Constant Bit Rate

Appendix

Random Integer Generator Generate

Random Integer Generator Generate integers randomly distributed in the range [0, M-1] Library Data Sources sub library of Comm. Sources Description The Random Integer Generator block generates uniformly distributed random integers in the range [0, M-1], where M is the M-aryl number defined in the dialog box. The M-aryl number can be either a scalar or a vector. If it is a scalar, then all output random variables are independent and identically distributed (i.i.d.). If the M-ary number is a vector, then its length must equal the length of the Initial seed; in this case each output has its own output range. If the Initial seed parameter is a constant, then the resulting noise is repeatable. Attributes of Output Signal The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See Signal Attribute Parameters for Random Sources in Using the Communications Block set for more details. The number of elements in the Initial seed parameter becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed parameter becomes the shape of a sample-based two-dimensional out put signal.

Integer-Input RS Encoder

Integer-Input RS Encoder Create a Reed-Solomon code from integer vector data Library Block sub library of Channel Coding Description The Integer-Input RS Encoder block creates a Reed-Solomon code with message length K and codeword length N. You specify both N and K directly in the block mask. The symbols for the code are integers between 0 and 2M-1, which represent elements of the finite field GF (2M). Restrictions on M and N are described in the section Restrictions on M and the Codeword Length N below. The difference N - K must be an even integer. The input and output are integer-valued signals that represent messages and codeword, respectively. The input must be a frame-based column vector whose length is an integer multiple of K. The output is a frame-based column vector whose length is the same integer multiple of N. For more information on representing data for Reed-Solomon codes, see the section Integer Format (Reed-Solomon only)." The default value of M is the smallest integer that is greater than or equal to log2 (N+1), that is, ceil (log2 (N+1)). You can change the value of M from the default by specifying the primitive polynomial for GF (2M), as described in the section specifying the Primitive Polynomial following. If N is less than 2M-1, the block uses a shortened Reed-Solomon code. An (N, K) Reed-Solomon code can correct up to floor ((N-K)/2) symbol errors (not bit errors) in each codeword. Specifying the Primitive Polynomial You can specify the primitive polynomial that defines the finite field GF (2M), corresponding to the integers that form messages and codeword. To do so, first check the box next to Specify primitive polynomial. Then, in the Primitive polynomial field, enter a binary row vector that represents a primitive polynomial over GF (2) of degree M, in descending order of powers. For example, to specify the polynomial, enter the vector [1 0 1 1]. If you do not select the box next to Specify primitive polynomial, the block uses the default primitive polynomial of degree M = ceil (log2 (N+1)). You can display the default polynomial by entering pimply (ceil (log2 (N+1))) at the MATLAB prompt. Restrictions on M and the Codeword Length the restrictions on the degree M of the primitive polynomial and the codeword length N are as follows: If you do not select the box next to Specify primitive polynomial, N must lie in the range. If you do select the box next to Specify primitive polynomial, N must lie in the range and M must lie in the range. Specifying the Generator Polynomial You can specify the generator polynomial for the Reed-Solomon code. To do so, first select the box next to Specify generator polynomial. Then, in the Generator polynomial field, enter an integer row vector whose entries are between 0 and 2M-1. The vector represents a polynomial, in descending order of powers, whose coefficients are elements of GF (2M) represented in integer format. See the section Integer Format (Reed-Solomon only) for more information about integer format. The generator polynomial must be equal to a polynomial with a factored form where is the primitive element of the Galois field over which the input message is defined, and b is an integer. If you do not select the box next to Specify generator polynomial, the block uses the default generator polynomial, corresponding to b=1, for Reed-Solomon encoding. You can display the default generator polynomial by typing rsgenpoly (N1, K1), where N1 = 2M-1 and K1 = K+ (N1-N), at the MATLAB prompt, if you are using the default primitive polynomial. If the Specify primitive polynomial box is selected, and you specify the primitive polynomial specified as poly, the default generator polynomial is rsgenpoly (N1, K1, and poly).

Spectrum Scope

Spectrum Scope Compute and display the short-time FFT of each input signal. Library's Sinks Description the Spectrum Scope block computes and displays the magnitude-squared FFT of the input. The input is a 1-D vector or a 2-D matrix of any frame status. When the input is a 1-by-N sample-based vector or M-by-N sample-based matrix, you must select the Buffer input check box. Each of the N vector elements (or M*N matrix elements) is then treated as an independent channel, and the block buffers and displays the data in each channel independently. When the input is frame-based, you can leave the input as is, or rebuffed data by checking the Buffer input check box and specifying the new buffer size. In the latter case, you can also specify an optional Buffer overlap. Buffering 1-D vector inputs is recommended. In this case, the inputs are buffered into frames (the length of which are specified in the Buffer size parameter), where each 1-D input vector becomes a row in the buffered outcome. If a 1-D vector input is left unbuffered, you will get a warning because the block is computing the FFT of a scalar; though the scope window appears, it is unlikely you will be able to see the plot, and a warning is also displayed on the scope itself. It is not recommended that you leave 1-D inputs unbuffered. The number of input samples that the block buffers before computing and displaying the magnitude FFT is specified by the Buffer size parameter, Mo. The Buffer overlap parameter, L, specifies the number of samples from the previous buffer to include in the current buffer. The number of new input samples the block acquires before computing and displaying the magnitude FFT is the difference between the Buffer size and Buffer overlap, Mo-L. The display update period is (Mo-L)*Ts, where Ts is the input sample period, and is equal to the input sample period when the Buffer overlap is Mo-1. For negative Buffer overlap values, the block simply discards the appropriate number of input samples after the buffer fills, and updates the scope display at a slower rate than the zero-overlap case. When the FFT length check box is cleared and the input is buffered, the block uses the buffer size as the FFT size. If the check box is cleared and the input is not buffered, the block uses the input size as the FFT size. When the check box is selected, the FFT length parameter, Nfft, is enabled, and specifies the number of samples on which to perform the FFT. The block zero pads or truncates every channel's buffer to Nfft before computing the FFT. The number of spectra to average is set by the Number of spectral averages parameter. Setting this parameter to 1 effectively disables averaging; See Short-Time FFT for more information. In order to correctly scale the frequency axis (i.e., to determine the frequencies against which the transformed input data should be plotted), the block needs to know the actual sample period of the time-domain input. This is specified by the Sample time of original time series parameter, Ts. When the Inherit sample time from input check box is selected, the block computes the frequency data from the sample period of the input to the block. This is valid when the following conditions hold: The input to the block is the original signal, with no samples added or deleted (by insertion of zeros, for example). The sample period of the time-domain signal in the simulation is equal to the period with which the physical signal was originally sampled. One example when these conditions do not hold, is such as when the input to the block is not the original signal, but a zero-padded or otherwise rate-altered version. In such cases, you should specify the appropriate value for the Sample time of original time-series parameter. The Frequency unit's parameter specifies whether the frequency axis values should be in units of Hertz or rad/s, and the Frequency range parameter specifies the range of frequencies over which the magnitudes in the input should be plotted. The available options are [0...Fs/2], [-Fs/2...Fs/2], and [0...Fs], where Fs is the time-domain signal's actual sample frequency. If the Frequency unit's parameter specifies Hertz, the spacing between frequency points is 1/ (Nifty). For Frequency units of rad/sec, the spacing between frequency points is 2/ (Nifty). Note that all of the FFT-based blocks in the DSP Block set, including those in the Power Spectrum Estimation library, compute the FFT at frequencies in the range [0,Fs). The Frequency range parameter controls only the displayed range of the signal. For information about the scope window, as well as the Display properties, Axis properties, and Line properties panels in the dialog box, see the reference page for the Vector Scope block.

Integer-Output RS Decoder

Integer-Output RS Decoder Decode a Reed-Solomon code to recover integer vector data Library Block sub library of Channel Coding Description The Integer-Output RS Decoder block recovers a message vector from a Reed-Solomon codeword vector. For proper decoding, the parameter values in this block should match those in the corresponding Integer-Input RS Encoder block. The Reed-Solomon code has message length K and codeword length N. You specify both N and K directly in the block mask. The symbols for the code are integers between 0 and 2M-1, which represent elements of the finite field GF (2M). Restrictions on M and N are described in the section Restrictions on M and the Codeword Length N following. The difference N - K must be an even integer. The input and output are integer-valued signals that represent messages and codeword, respectively. The input must be a frame-based column vector whose length is an integer multiple of K. The output is a frame-based column vector whose length is the same integer multiple of N. For more information on representing data for Reed-Solomon codes, see the section Integer Format (Reed-Solomon only)." The default value of M is the smallest integer that is greater than or equal to log2 (N+1), that is, ceil (log2 (N+1)). You can change the value of M from the default by specifying the primitive polynomial for GF (2M), as described in the section specifying the Primitive Polynomial below. If N is less than 2M-1, the block uses a shortened Reed-Solomon code. You can also specify the generator polynomial for the Reed-Solomon code, as described in the section specifying the Generator Polynomial. An (N, K) Reed-Solomon code can correct up to floor ((N-K)/2) symbol errors (not bit errors) in each codeword. The second output is the number of errors detected during decoding of the codeword. A -1 indicates that the block detected more errors than it could correct using the coding scheme. An (N, K) Reed-Solomon code can correct up to floor ((N-K)/2) symbol errors (not bit errors) in each codeword. You can disable the second output by clearing the box next to Output port for number of corrected errors. This removes the block's second output port. The sample times of the input and output signals are equal.

Integer to Bit Converter

Integer to Bit Converter Map a vector of integers to a vector of bits Library Utility Functions Description the Integer to Bit Converter block maps each integer in the input vector to a group of bits in the output vector. If M is the Number of bits per integer parameter, then the input integers must be between 0 and 2M-1. The block maps each integer to a group of M bits, using the first bit as the most significant bit. As a result, the output vector length is M times the input vector length. The input can be either a scalar or a frame-based column vector.

Error Rate Calculation

Error Rate Calculation Compute the bit error rate or symbol error rate of input data LibraryComm Sinks Description the Error Rate Calculation block compares input data from a transmitter with input data from a receiver. It calculates the error rate as a running statistic, by dividing the total number of unequal pairs of data elements by the total number of input data elements from one source. You can use this block to compute either symbol or bit error rate, because it does not consider the magnitude of the difference between input data elements. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate. This block inherits the sample time of its inputs. Input Data this block has between two and four input ports, depending on how you set the mask parameters. The imports marked TX and Rx accept transmitted and received signals, respectively. The TX and Rx signals must share the same sampling rate. The TX and Rx inputs can be either scalars or frame-based column vectors. If TX is a scalar and Rx is a vector, or vice-versa, then the block compares the scalar with each element of the vector. (Overall, the block behaves as if you had preprocessed the scalar signal with the DSP Block set's Repeat block using the Maintain input frame rate option.) If you check the Reset port box in the mask, then an additional import appears, labeled Rst. The Rst input must be a sample-based scalar signal and must have the same sampling rate as the TX and Rx signals. When the Rst input is nonzero, the block clears its error statistics and then computes them anew. If you set the Computation mode mask parameter to select samples from port, then an additional inport appears, labeled Sel. The Sel input indicates which elements of a frame are relevant for the computation; this is explained further, in the last sub bullet below. The Sell input can be either a sample-based column vector or a one-dimensional vector. The guidelines below indicate how you should configure the inputs and the mask parameters depending on how you want this block to interpret your TX and Rx data. If both data signals are scalar, then this block compares the TX scalar signal with the Rx scalar signal. You should leave the Computation mode parameter at its default value, Entire frame. If both data signals are vectors, then this block compares some or all of the TX and Rx data: If you set the Computation mode parameter to Entire frame, then the block compares the entire TX frame with the entire Rx frame. If you set the Computation mode parameter to select samples from mask, then the selected samples from frame field appears in the mask. This parameter field accepts a vector that lists the indices of those elements of the Rx frame that you want the block to consider. For example, to consider only the first and last elements of a length-six receiver frame, set the Selected samples from frame parameter to [1 6]. If the Selected samples from frame vector include zeros, then the block ignores them. If you set the Computation mode parameter to select samples from port, then an additional input port, labeled Sell, appears on the block icon. The data at this input port must have the same format as that of the selected samples from frame mask parameter described above. If one data signal is a scalar and the other is a vector, then this block compares the scalar with each entry of the vector. The three sub bullets above are still valid for this mode, except that if Rx is a scalar, then the phrase "Rx frame" above refers to the vector expansion of Rx.