Advanced Communications Project

Technology Reference Document

Prepared for

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EXECUTIVE SUMMARY

This document is an updated version of the 1985 *Technology Reference Document* prepared for the U.S. Coast Guard Research and Development Center. It has fully incorporated the contents of the 1985 document. No attempt was made to perform any editorial changes on the 1985 material, with the sole exception of correcting any information which is no longer accurate or applicable due to the evolution of technology. The major changes to the document have been in the area of reorganization / style, the addition of new telecommunications technology information, and the inclusion of information regarding the U.S. navy Satellite Communications System.

To provide an all encompassing technology document was beyond the scope of this particular effort. Consequently, the new technology introduced with this updated document has focused on selected topics which were considered by the authors to be of prime importance to U.S. Coast Guard personnel. The following technologies were selected:

- Electromagnetic Frequency Spectrum
- Transmission Hardware Technology
- Over the Horizon Radio Communications
- Analog Modulation
- Digital Modulation
- Radio Frequency Modulation
- Information Coding/Processing
- Communications Channel Multiplex/ Access
- Communications Networks
- Integrated Services Digital Network
- Cellular Mobile Transmission
- Satellite Communications
- Computer Architecture Concepts
- Security
- Communications and Network Services
- Fleet Satellite Communications
- U.S. Navy Users and Network Architecture
- Interoperability Architecture

The new topics added to this edition of the *Technology Reference Document* were arbitrarily selected based on industry and DoD trends and technology feasibility. Since many areas of telecommunications technology are undergoing rapid evolution, comments are anticipated and are actively solicited for eventual inclusion into future editions of this document.

1 ELECTROMAGNETIC FREQUENCY SPECTRUM

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1.1 Frequency Spectrum Definition

The continuum of the electromagnetic frequency spectrum useful for communications range from the Extremely Low Frequency (ELF) (designated 30-300 hertz) to the Extremely High Frequency (EHF) (designated 30-300 GHz) millimeter waves. Table 1-1 below specifies the designated nomenclature for those frequencies used in communications.

Table 1-1. Electromagnetic Spectrum Nomenclature

FREQUENCY RANGE	DESIGNATION	CLASSIFICATION
30 – 300 Hz	ELF	Extremely Low Frequency
3 – 30 kHz	VLF	Very Low Frequency
30 – 300 kHz	LF	Low Frequency
.3 – 3 MHz	MF	Medium Frequency
.3 – 30 MHZ	HF	High Frequency
30 – 300 MHz	VHF	Very High Frequency
. 3 – 3 GHz	UHF	Ultra-high Frequency
3 – 30 GHz	SHF	Super High Frequency
30 – 300 GHz	EHF	Extremely High Frequency

Letter designations are also used for Radio Frequency (RF). Table 1-2 lists the military letter designations for RF frequencies.

Table 1-2. Military Radio Frequency Bands

BAND DESIGNATION	FREQUENCY RANGE
A	0 – 250 MHz
B	250 – 500 MHz
C	500 – 1000 MHz
D	1 – 2 GHz
E	2 – 3 GHz
F	3 – 4 GHz
G	4 – 6 GHz
H	6 – 8 GHz
I	8 – 10 GHz
J	10 – 20 GHz
K	20 – 40 GHz
L	40 – 60 GHz
M	60 – 100 GHz

Electromagnetic waves are propagated from a transmitting antenna to a receiving antenna in a number of different ways besides through a direct path through the atmosphere (line-of-sight/space wave). They may also be propagated along the earth's surface (ground wave), or through refraction or reflection or scattering from natural atmospheric reflectors. Significant reflecting and scattering modes of electromagnetic propagation include ionospheric and tropospheric scattering. Each of these types of electromagnetic propagation is further discussed in Section 3 *(Over-the-Horizon Radio Communications).* Figure 1-1 illustrates the various types of electromagnetic propagation.

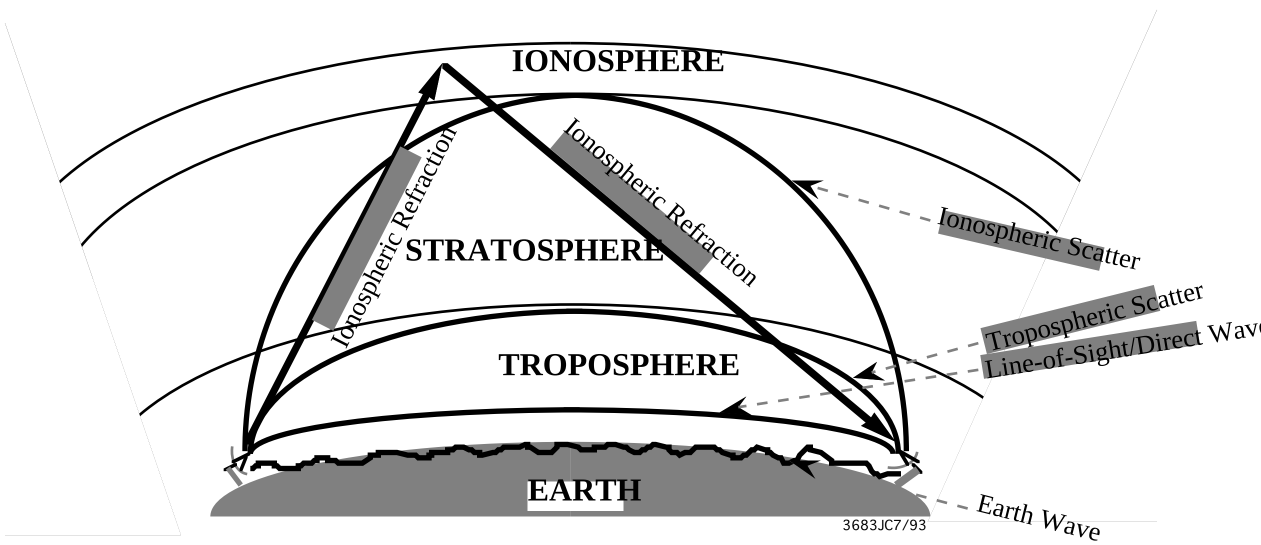


Figure 1-1. Electromagnetic Propagation

1.2 Extremely Low Frequency (ELF)

ELF is the communications band extending from 30 Hz to 300 Hz. Transmission is conducted over areas of low conductivity bedrock. The signal is broadcast by orthogonal antennas strung horizontally over the rock formation terminating in low resistance ground terminals at either end. The current flows into the rock and because of the low conductivity of the medium, the signals are driven deep into the earth from one terminal to the other. This effectively produces a loop antenna. The signal radiates into the atmosphere, traveling between the ionosphere and the earth's surface. The signal penetrates the surface of the ocean and propagates to great depths. Clearly, ELF is an attractive transmission mechanism for communicating with submarines. In fact, submarines are the primary ELF communications users. ELF systems decrease the vulnerability of our submarine force as communications are conducted at patrol depths and speeds. ELF possesses two key advantages:

- Efficient signaling schemes and sophisticated receiver designs help minimize the required transmitter power. Very low radiated power (2 to 8 watts) is required to transmit at great distances.
- ELF signals are inherently resistant to the effects of electromagnetic pulses.

Nevertheless, there are substantial drawbacks with ELF communications due to factors such as:

- Continual setbacks in developing ELF communications occur due to political and environmental debates concerning the potential health hazards although no health or environmental hazards have been substantiated.
- ELF transmitting antennas require a large ground plane. Proposed ELF sites will require 134 km of buried antenna. The system cost is both expensive and geographically expansive.
- Site selection and antenna design process are always time consuming and have cost factors with which to contend.
- The electromagnetic fields produced by ELF transmitter sites may cause interference problems to local power distribution lines and telephone circuits.
- ELF communications systems are susceptible to many forms of jamming although signal bandwidth spreading can alleviate the effects.
- Data rate transmission is slow.

ELF systems, as used today, are intended primarily as a wartime submarine communications system. We can expect to see continued engineering developments in tactical communications with submarines, particularly in the following technical areas:

- Propagation phenomena (attenuation, diurnal and seasonal variation weather)
- Signal processing in a noise environment
- Message pre-formatting and coding techniques due to a slow data rate
- Signal bandwidth spreading in order to protect against hostile Electronic Counter Measures (ECM)
- Synchronous transmitter operation to increase signal strength and reliability

1.3 Very Low Frequency (VLF)

VLF is the communications band extending from 3 kHz to 30 kHz. VLF radio waves are used for long range communications and navigation. Wave propagation is accomplished via ducting between the earth's surface and the lower ionosphere boundary. VLF has limited use in transmission to points below the sea's surface and on the earth's surface. An important factor in VLF transmission is ground loss resistance. The amount of resistance varies inversely with the total length of wire in the ground system. Hundreds of kilometers of wire are needed to keep ground resistance down to tenths of an ohm. Antennas are vertically polarized. It does however permit world-wide coverage and is relatively insensitive to physical obstacles. Furthermore, propagation is the same over both land and water. However, it requires a very large ground plane and transmitting antenna which characteristically have very low gain. Equipment is typically large and expensive. Also, a large land area is required for the ground plane. This transmission frequency has, nevertheless, been used in Coast Guard applications – the Omega Navigation System operates in the VLF band. The technology required to further use this radio frequency will continue with the substantial ongoing work being done in the field of submarine communications.

1.4 Low Frequency (LF)

Low frequency (30 kHz to 300 kHz) are propagated to distances of a few hundred kilometers by direct-ray diffraction (surface waves) around the curvature of the earth and by indirect wave reflection from the ionosphere. LF systems are normally used for intermediate ranges of 1000 to 5000 km.

Surface wave propagation studies indicate that in the LF through HF frequency ranges, field strengths increase with decreasing frequency and are stronger over sea water than over land. Vertical polarization gives higher field strengths than horizontal.

Sky waves cause fading at 80 to 800 km distances and generally produce higher-level field strengths than the surface wave at longer distances. Sky wave propagation is influenced greatly by seasonal, diurnal, and irregular variations due to the changing ionospheric properties (LORAN-C difficulties). Phase variation, due to direct and indirect ray interference, can upset time and distance measurements. Diurnal variations of 10 dB or more in received field strengths have been observed.

At great distances, LF communications are inadequate for telephone transmission and is marginal even for telegraphic transmission. However, if bandwidth reductions are made, the signal-to-noise ratio (E_b/N) is noticeably improved. With noise limited LF communications systems world-wide coverage is obtained. Modulation schemes primarily used are on-off keying and Frequency Shift Keying (FSK). LF communications have several advantages over the shorter-wavelength frequencies. For instance, they exhibit:

- Much better propagation uniformity
- An ability to be relatively independent of terrain and obstacles
- The ability to penetrate deep into the ground and several meters into the sea
- The ability to provide stable transmission over distances up to about 1500 km

LF communications, however, require high power and complex antennas for long range communications. Furthermore, at transatlantic distances the atmospheric noise level and signal field strength both increase with increasing wavelength. This makes the choice of operating frequency for communications systems somewhat uncertain. We can expect communications over this frequency to be applied to Coast Guard requirements such as for:

- Radio beacons for transmitting Differential Loran-C or Global Positioning System (GPS) corrections
- Radio Direction Finding (RDF) calibration
- Very low speed data communications channels

In the future, we can anticipate further engineering work in applying this communications frequency as an adjunct tool in the area of navigation.

1.5 Medium Frequency (MF)

The Medium Frequency band ranges from 300 kHz to 3 MHz. Signals propagate by both ground and sky waves. MF telegraphy provides adequate marine communications up to 650 km. MF voice can be used up to 250 km with some success. Ground wave characteristics in the MF band include:

- No fading of the signal with time
- No de-correlation between different frequencies
- No de-correlation between vertical and horizontal polarization

Sky wave propagation at MF is primarily limited to the nighttime because of high absorption in the ionosphere during daytime. Occasionally, severe fading and multi-path effects are experienced as ground waves and sky waves collide in what are called "transition regions". Sky wave propagation becomes more dominant over ground waves with distance and is known to be limited by

- Absence of signals during the daytime due to absorption

- Atmospheric noise (greater at night)
- Signal fade-outs

The most serious drawbacks to the use of the MF communications are the facts that the MF band is overcrowded and that severe fading and multipath can disrupt viable communications. Nevertheless, MF communications is still being applied to the Coast Guard's voice and telegraphy communications systems and provides differential corrections for GPS or Loran-C Systems.

Medium frequency radio is used by the Coast Guard for marine safety broadcast, distress signaling, and marine weather information. From selected Coast Guard stations, medium frequency broadcast coverage can reach out approximately 70 nautical miles. Continuous Wave (CW) signals are used to trigger radio alarm signals. CW signals are broadcast at a frequency range of approximately 420 to 540 kHz. One system utilizing the CW signals is the Automated Mutual-Assistance Vessel Emergency Rescue (AMVER) system. Voice transmissions occur at the international standard of 2182 kHz.

1.6 High Frequency (HF)

High Frequency (HF) radio transmission is used substantially for long distance communications. It is an extremely cost effective (inexpensive widely available technology) radio frequency for implementing long-range communications. HF is defined to be the frequency range 3-30 MHz, although frequencies down to 1.5 MHz are often considered part of the HF band. Propagation is by ground and sky wave. The ground wave is usually attenuated sufficiently to be unusable after several hundred kilometers. Sky wave propagation depends on the different layers of the ionosphere, and therefore goes through diurnal, seasonal and sunspot cycles, as well as being affected by latitude. Sky wave propagation can offer global coverage under optimum conditions, although it is susceptible to magnetic and electrical disturbances such that coverage is not always reliable. It is also quite susceptible to multipath effects. Furthermore, it is not unusual for the sky wave to return to earth intermittently over distance, producing skip zones where reception is impossible. In the areas where both ground wave and sky wave can be received, the two will most likely be received out of phase due to the different propagation path lengths. The greatest disadvantage of HF communications steps from its universal popularity and use – only a narrow bandwidth is allowed per channel and crowded HF bands often require frequency changes to avoid interference or to accommodate the above described cyclic changes. Furthermore, frequency agility required for short or long term reliability and the current standard technology requires a very low baud rate to reduce multipath effects. The current trend in the evolution of HF technology is the development and perfection of digital transmission techniques. These include and are not limited to real-time channel evaluation techniques, specialized modems and error correction. Real-time channel evaluation is implemented by "adaptive" radios which select a "quality" channel by actual on-line testing. The most complete system consists of a Chirp Sounder, a transmitter ashore that sweeps the HF with a unique signal and a receiver afloat that measures the receive signal and determines the best frequency to use in just a few minutes. The Chirp Sounder also acquires reference data for propagation prediction models. New modems frequently appear in the competitive marketplace. Automatic Repeat Request (ARQ) is being supplanted by Time and Diversity modems as well as HF packet switching modems. State-of-the-art error correcting/detecting is becoming commonplace with today's HF technology. It can be anticipated that future developments error detection/correction will exploit advances in academia.

1.7 Very High Frequency (VHF)

Very High Frequency (VHF) radio waves are those from 30 MHz to 300 MHz although tactically it is normally considered to be below 225 MHz. VHF propagation is by line-of-sight (LOS), reflections from the terrain, man-made structures, reflections re-radiation from meteor trails (meteor burst), and tropospheric scatter. Other than meteor burst and tropospheric scatter, power is generally limited to 150 watts and the range extends to LOS (although there are documented cases of systems separated by 320 km communicating over VHF is ducting environments). VHF is currently used for voice, data, telemetry, and television applications. VHF systems are reliable and very economical. It requires very small antennas and the necessary antenna gain is very easily achieved. However, as noted above, it is generally limited to LOS. Ranges well beyond LOS are only achievable at frequencies below 160 MHz with over water transmission paths (large terrestrial obstacles such as hills and tall buildings create shadows and odd reflective patterns). For the Coast Guard, VHF communications can be used for any type of short-range communications including voice and computer-to-computer synchronization communications. It is anticipated that this technology will continue to be improved. Specifically, much work has been and will continue to be expended in the areas of digitized voice and coding techniques to minimize overall channel bit error rates.

1.8 Ultrahigh Frequency (UHF)

Ultrahigh Frequency (UHF) radio transmission is from 300 MHz to 3000 MHz (3 GHz), although 225 to 400 MHz is the primary military UHF communications band. UHF is propagated LOS, allowing use of the same frequencies within comparatively small distances. UHF can be used for fixed, point-to-point links using moderate gain, directed antennas. Propagation is affected by multipath effects and bending of the beam by variations in the refractive index. It is also extensively used by satellite communications systems (particularly the U.S. Navy). For LOS applications the available UHF technology is both reliable and inexpensive and requires very low power. However, it is very susceptible to interference from large obstacles such as hills and tall buildings. Furthermore, the upper end of the frequency band is adversely affected by heavy rain (absorption). Nevertheless, its utility for the Coast Guard will continue for aviation, marine communications, and mobile communications with satellites.

1.9 Super high Frequency (SHF)

The Super high Frequency (SHF) band (3 GHz–30 GHz) contains the so-called *microwave* frequencies and features wavelengths ranging from 1 to 10 cm (*centimeter waves*). The predominant difference between UHF and SHF characteristics is the increased absorption, scattering, and attenuation due to moisture in the signal's path. Radio-wave attenuation by atmospheric gases and water, and multipath effects become increasingly significant above 10 GHz.

Applications for SHF propagation is twofold. Terrestrial LOS transmission and earth-space communications links are possible, and in wide usage today. With regard to terrestrial LOS communications at frequencies above 10 GHz, multipath can limit the bandwidth to a few tens of megahertz. Additionally, selective fading is of paramount concern. Nevertheless, diversity techniques aid in minimizing the frequency and duration of atmospheric multipath fading in LOS systems, as well as earth-space communications links. Earth-space communications links consider a value of 20 GHz to be the maximum usable frequency for ground-space communications due to atmospheric attenuation. In rainy climates, 10 GHz is recommended as the maximum. Usage of SHF today include satellite communications, radar microwave links, and experimental television.

Currently, there exists a great amount of research and development of earth-space SHF communications system technology focusing on minimizing atmospheric losses and expanding bandwidth capabilities. However, research and development has also been focused in the area of LOS applications with possible direct Coast Guard applications. For instance work has been performed in attempting to extend the maximum distances for beyond LOS communications for marine vessels utilizing signal enhancements provided by evaporation ducting (vertical gradient in refractive index primarily caused by variation in humidity).

1.10 Light Amplification by Stimulated Emission of Radiation (LASER)

LASER components convert electrical energy into visible spectrum radiation. This energy, unlike conventional light energy, is highly monochromatic, and temporally and spatially coherent. Temporal coherence is the energy waves' property of traveling at the same frequency and perfectly in-step (in-phase) with the stimulating radiation.

Spatial coherence refers to the energy waves traveling in the same direction and having the same phase across any one of the wave fronts. Today, in fiber optic communications systems, LASER diodes are used to generate the optical signals. The use of lasers in communications may be divided into four general application categories:

- Terrestrial short-range paths through the atmospheres
- Closed-pipe systems for sending data at high rates between and within mayor metropolitan centers
- Near-space communications for relaying data at high rates
- Deep-space communications from the outer planets

There are two basic types of LASER communication receiver systems. Heterodyne communications receiver systems (photon mixing) and direct detection. With the heterodyne communications system, the LASER signal is modulated, either by orthogonal phase-shift modulation, Frequency Modulation (FM), or Pulse-Code Modulation (PCM). The received signal is mixed with a local oscillator, and the beat frequency passes through the Intermediate Frequency (IF) to a second detector, where the signal is recovered. In direct detection, the received LASER energy is collected and focused on a photo detector which responds only to the signal's intensity changes. lasers provide many inherent advantages to communications applications. These include:

- Directivity attainable with small antennas
- Inherent and easily used wide bandwidths
- Power required for transmission much less than that associated with microwave transmissions (primarily due to its directionality characteristics)
- Overlapping of spectrums for different links without interference problems (due to its directionality characteristics)

LASER technology for communications applications does, nevertheless, possess some significant drawbacks. These include:

- Problems with signal-quantum noise and background noise
- Stringent spatial requirements are necessary when photon mixing is employed (e.g., the atmosphere can easily corrupt the phase front, thus making it difficult to perform heterodyning)
- Quantum noise at optical frequencies become significant when compared to noise in the microwave region – quantum noise increases linearly with frequency

Future LASER communications, weapons, and/or navigation systems for the Coast Guard are anticipated as LASER technology is further developed and implemented within the world of commercial telecommunications and the military. A need for high-speed-data communications and advanced weapon systems will continue to exist. Continued improvements based on LASER technology can be expected for both land line and short-range terrestrial communications applications. It is perceived that LASER communications within the Coast Guard is probable in the very near future. In addition to the LASER technology applications for fiber optic communications, we can expect LASER technology applied for Coast Guard applications such as navigation systems. For instance, lasers for Single Station Range Lighting is a proposed solution for the problems of closer-to-shore navigation with background lighting. Fiber optic based land-line (as well as onboard afloat platforms) and short-range LASER Low Probability of Detection (LPD) communications systems are key areas in which this technology will be of great use to the Coast Guard. LASER technology is a very significant communications technology for the Coast Guard. It can be expected that LASER based systems will become even more economical, more readily available, and provide significant capabilities for current and future Coast Guard communications requirements.

2 TRANSMISSION HARDWARE TECHNOLOGY

2.1 Open Wire

This is a simple physical transmission circuit used for half-duplex data communications also called "open pair". It consists of two conducting wires separated by an insulator. This circuit is balanced (both conductors have equal impedance relative to ground). Open wire circuits are extremely susceptible to noise (cross-talk, electromagnetic interference, etc.) and feature radiation loss which is directly proportional to signal frequency. Consequently, open wires are generally used for transmitting low-frequency signals in a relatively "noiseless" environment.

2.2 Waveguides

Waveguides are transmission "lines" created from a hollow conductor within which radio waves are propagated between two locations. They feature extremely low attenuation at microwave frequencies and are used at RF installations processing signals above 2 GHz to interconnect antenna with radio equipment. Waveguides must be specifically designed and manufactured according to the separation of the antenna and radio equipment as well as the electromagnetic frequency band which is desired to be transmitted. Three main types of waveguides are used: 1) circular, 2) elliptical, and 3) rectangular.

In order to achieve these attenuation characteristics, waveguides must be manufactured with precise uniformity. In addition, its implementation requires careful installation analysis and design in order to achieve the maximum possible transmission efficiency. For instance, although circular waveguides feature the lowest possible attenuation, they are only useful in straight-line rigid configurations. Thus, the less efficient "bend-able" rectangular and elliptical waveguides are commonly used due to their utility over long continuous distances and through tight spaces. To optimize a waveguide design, the designer must carefully seek to minimize attenuation which is directly proportional to the number of waveguide components, bends, and joints.

2.3 Twisted Pair

A common physical transmission media in data communications is the so-called twisted pair. It consists of two insulated wires twisted together in a spiral serpentine manner in order to reduce the induction of external interference, including other twisted pairs in close proximity. Typically, they are constructed with a pair of very thin insulated copper wires and are used for both analog and digital communications. The digital bandwidth provided by twisted pairs are dependent on the thickness of the wires and are generally on the order of megabits per second. Twisted pairs can generally be run for a few kilometers without amplification repeaters.

2.4 Coaxial Cable

This type of transmission line is constructed of an inner conductor surrounded by an outer conductor. The conductors themselves are separated by a dielectric insulator. Due to the physical construction of the coaxial cable, it provides a bandwidth much higher than that provided by twisted pairs and much higher frequencies may be passed over it. Coaxial cables are characterized by their impedance values. Figure 2-1 illustrates the parts of the coaxial cable.

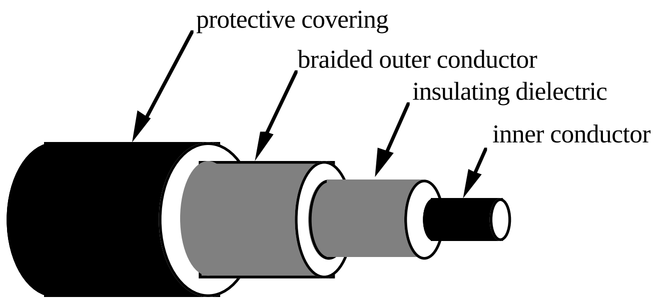


Figure 2-1. Coaxial Cable

The two most common types of coaxial cables in use today are the 50 ohm ("baseband") and 75 ohm ("broadband") varieties. The broadband variety is used for analog transmission. Its primary application has been in the area of cable television. Broadband networks can be used to transfer digital signals by converting the digital information into analog signals or vice-versa for the recipient of the information. However, due to analog signaling, a broadband coaxial cable can support a aggregate data rate of up to 200 megabits per second.

The bandwidth obtained from a baseband coaxial cable is strongly dependent on the length of the cable. A data rate of 10 megabits per second can be supported on cable lengths up to 1 kilometer.

2.5 Fiber Optics

This is the transmission technology in which information is transferred through thin strands of glass on a modulated optical beam. The basic system consists of an optical source, an optical fiber, and an optical detector

Fibers that are used for optical communications are waveguides made of transparent dielectric whose function is to guide visible and infrared light over long distances. An optical fiber consists of an inner cylinder of glass called the core, surrounded by a cylindrical shell of glass or plastic of lower refractive index, called the cladding. The basic premise of this transmission technology is the property of glass which causes light to be refracted and dispersed predictably. The outer layer of the glass conduit is a substance (cladding) which has a much higher refracting characteristic than the core which prevents the light from leaving the fiber. Figure 2-2 illustrates these concepts.

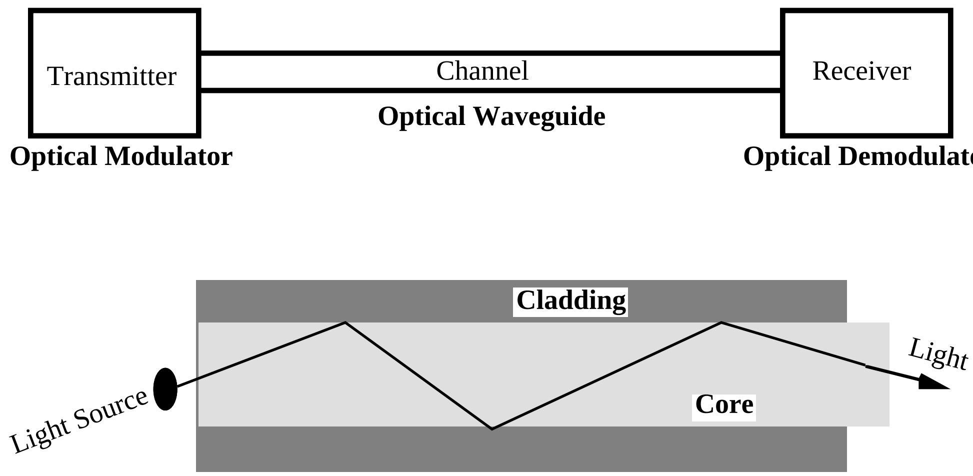


Figure 2-2. Fiber Optic System

An optical fiber communications system consists of a signal (speech) being converted into electrical signals in the transmitter, which then modulates the optical intensity of the source, such as a semiconductor LASER or light emitting diode. The optical signals are transmitted through the optical fiber waveguide using the properties of electromagnetic waves-ray model and Maxwell's equations. The optical signals are detected by a photo detector, which reconverts them into electrical signals. The electrical signals, such as PCM, are converted then into sound waves-speech. Figure 2-3 illustrates this sequence.

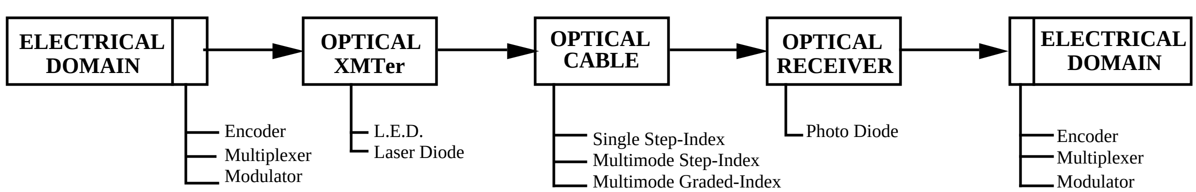


Figure 2-3. Fiber Optic Transmission Sequence

Optical fibers make better use of the advantages of light transmission than any other optical waveguide system. Optical fibers are classified into three types as summarized in Table 2-4.

Table 2-1. Optical Fiber Classification

TYPE	DESCRIPTION
Single Mode Step Index or Mono Mode Step Index	Very narrow core limiting wavelength which can pass through it. Low signal attenuation and dispersion. Suitable for long- haul high-bandwidth applications
Multimode Step Index	Large core. Generally used for low band-width short-range applications
Multimode Graded Index	Even larger core. Generally used for high-bandwidth medium-range applications

Fiber optic transmissions exhibit the following advantages:

- Low loss over a wide range of wavelengths (less than 1 dB/km)
- Large bandwidth (1 and 100 GHz, respectively, for multimode and single-mode fibers over 1 km)
- Reliability/durability (no corrosion or temperature problems)
- Signal security as optical fibers do not radiate energy
- No ground loops (i.e., no induced electromagnetic fields)
- Small size/low weight (a cladded glass fiber has a total diameter of about 100 mm and a total diameter, inclusive of the plastic coatings
- Small cross talk (no generation or conduction of electromagnetic noise or interference) and high security
- Natural abundance of glass materials, especially those containing a high concentration of silica
- High resistance to chemical attack and temperature variations

It does possess the following significant drawbacks:

- Signal attenuation (fiber loss or signal loss) is an area of major concern. It dictates repeater separation and, thus, impacts overall system cost. The attenuation mechanisms include absorption, scattering, and radiation losses of optical energy
- Regenerators separation and usage. Regenerators are expensive to fabricate, install, and maintain
- Signal distortion is a result of signals broadening as they travel along a fiber, thus creating errors in the receiver output. Material dispersion, waveguide dispersion and inter-modal dispersion contribute to a signal (pulse) spreading out in time

For the Coast Guard, clearly fiber optic technology will be of great use for intra-ship communications systems. For information systems, particularly in the area of office automation, their use in local area networks (LANs) will minimize the cable runs in offices as the cables are much smaller than its other copper counterparts. Its use in communications systems requiring transmission security is quite clear. It is anticipated, based on the evolution of the technology within the last decade, that we can anticipate ongoing improvements in the following areas:

- Reduction in signal degradation and attenuation
- Fiber materials and fabrication methods
- Optical sources such as Light Emitting Diodes (LED) and LASER Diodes
- Coupling and photo detectors

3 OVER-THE-HORIZON RADIO COMMUNICATIONS

3.1 Overview

3.2 Diversity and Fading

3.3 Ionospheric Scatter and Refraction Transmission

3.4 Troposcatter Transmission and Digital Troposcatter Transmission

3.5 Meteor Burst

3.1 Overview

Many techniques besides the use of modern satellites exist today for performing long range over-the-horizon radio communications. Techniques used include propagating radio waves along the earth's surface, through the atmosphere, and by reflecting the radio waves off natural reflectors. Techniques in use today include:

- HF communications. Uses ionospheric reflections for transmitting signals to distances of up to thousands of kilometers
- Meteor bursts. Uses reflections from the ionization produced by small meteors in the high atmosphere permitting the transfers of signals ranging up to thousands of kilometers
- Ionospheric scatter. Uses the ionospheric scattering of radio signals to transmit signals in the frequency range of 30-100 MHz ranging up to thousands of kilometers
- Troposcatter. Uses the non-homogeneous elements in the troposphere for transmitting signals over hops on the order of hundreds of kilometers

These techniques use the inherent characteristics of the atmosphere which consist of the following three zones:

- Troposphere. This is the zone closest to the ground extending to a height of about 10 km where clouds form and convection are predominating elements. The air is not ionized in this zone
- Stratosphere. This zone extends from the top of the troposphere to a height of about 50 km. This is a relatively still zone without the convection or perturbations of the troposphere. Humidity is nearly absent
- Ionosphere. This zone extends from the top of the stratosphere and features an appreciable amount of air ionization. The reflective properties of this region have been exploited for long range VHF transmissions (ionospheric scatter)

3.2 Diversity and Fading

The detrimental effects of atmospheric induced fading can be mitigated through the use of diversity systems. These systems offer an effective technique to improve long range communications. It has been shown that if two or more high frequency radio channels are sufficiently separated in space, frequency, angle of arrival, time, or polarization, the fading on the various channels is more or less independent. Diversity systems make use of this fact to improve the overall apparent signal quality by combining or selecting separate radio channels on a single high-frequency circuit. The most prevalent diversity reception techniques embody any or all of the following areas:

- Space diversity uses multiple receivers separated to combine received signals. When the signal at one receiver fades, there is a high probability that the S/N at some of the other receivers will be large so that the composite probability of error can be kept small. For optimum operation of a space diversity system, receivers must be separated by significant integral orders of wavelengths. Clearly, these diversity systems are only practical at large earth sites (HF feature wavelengths are in the 10 – 100 meter range)
- Time diversity uses the re-transmission of the message a number of times. The effect of a low received S/N at one time is counterbalanced by high received S/N at other times, thus providing a low composite probability of error
- Frequency diversity uses multiple frequency channels. When the signal fades in one channel, it is counterbalanced by a high received S/N in other channels

"Research indicates that satisfactory diversity improvement can be obtained if the correlation coefficient of the fading on the various channels does not exceed about 0.6. Experiments have indicated that a frequency separation of about 400 hertz gives satisfactory diversity performance on long high frequency paths. Spacing between antennas at right angles to the direction of propagation should be about 10 wavelengths. Polarization diversity has been found to be about equivalent to space diversity in the high frequency band. Measurements have indicated that times varying from 0.05 to 95 seconds may be necessary to obtain fading correlation coefficients as low as 0.6 in high frequency time diversity systems. Angle of arrival diversity requires large antennas so as to obtain the required vertical directive characteristics. Two degree differences in the angle of arrival have been shown to give satisfactory diversity improvement on high frequency circuits." (*Reference Data for Radio Engineers, 6th Edition*).

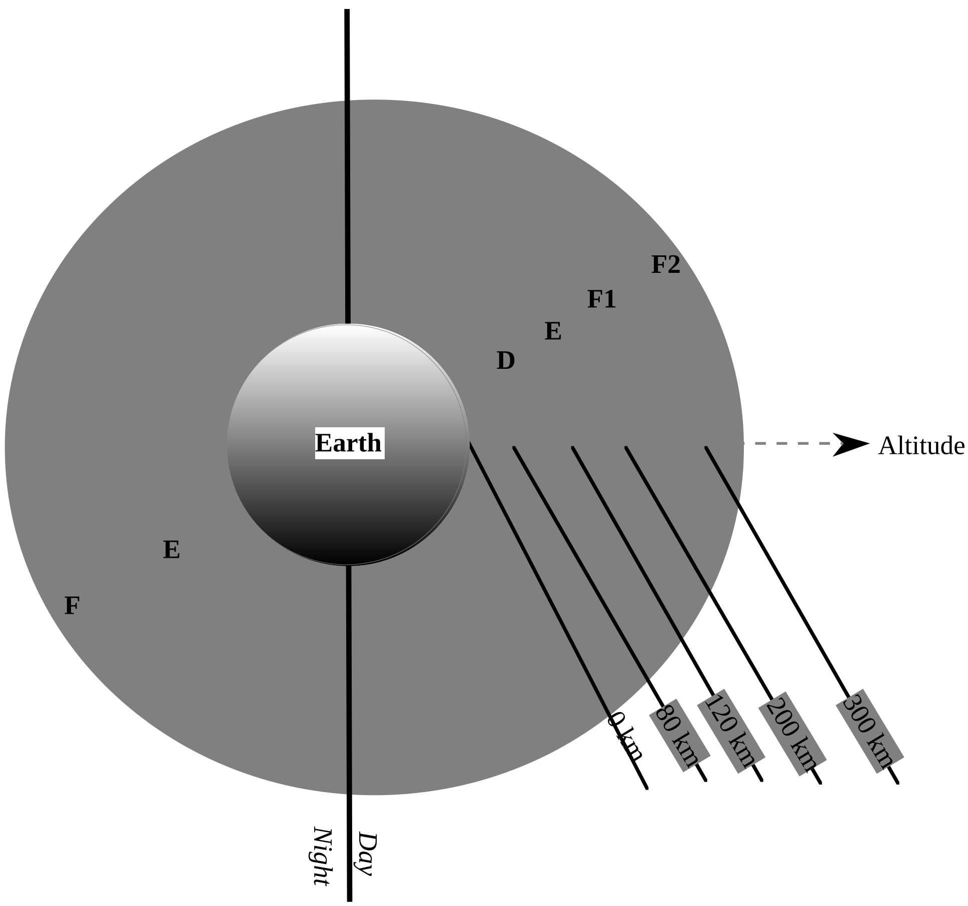
For the Coast Guard, these fading mitigation techniques offer a simple means to substantially improve HF and satellite communications. Significantly, time and frequency are simple to implement for HF applications. However, there are some drawbacks to the use of these fade mitigating techniques. They are:

- Inefficiency. Usage of a time diversity system is not an expedient technique for communications applications such as fleet broadcast. Much time can elapse by the time a verifiable message is obtained
- Uneconomical. Switching diversity systems (frequency diversity) requires the use of multiple radio receivers
- Noise. Switching of the receiver output signals may introduce information loss or transients into the external circuitry

We can expect to see further research into counter-measures with respect to multipath discursive fading. The various diversity techniques will further be investigated with respect to digital radio and multiple access methods and the correlation study of the various techniques.

3.3 Ionospheric Scatter and Refraction Transmission

Ionosphere skywave transmission exploits the significant ionization of the air which occurs in the ionosphere. This atmospheric region has been categorized into the four regions "D", "E", "F_1", and "F_2" as illustrated in Figure 3-1.



therefore this intensity is greatest directly under the sun. As ionization density varies directly with UV radiation, the density also varies with the seasons (i.e., sun's zenith angles). They differ, however, in that the "E" region effectively disappears after sunset whereas the "F_1" region combines with the "F_2" layer at sunset to form the "F" layer. The "F_2" region exists at all times. Its height varies directly with the intensity of solar heat and its properties in general are affected by the earth's magnetic field. These regions act as reflectors for HF propagation. The degree of reflection for the "E" and the "F" regions varies with the level of ionization. The refractive properties of these regions are employed for both single-hop and multi-hop skywave transmission. Successfully employing ionospheric refraction transmission between two over-the-horizon locations, therefore, requires careful understanding of the season, location latitudes, time-of-day, and the effects of atmospheric disturbances such as sunspots and magnetic storms.

3.4 Troposcatter Transmission and Digital Troposcatter Transmission

Troposcatter transmission exploits the irregularities in the troposphere to permit a site to communicate directly with another site beyond the horizon. Propagation over tropospheric paths is characterized as multipath transmission by reflection and/or refraction from many non-homogeneous elements within the common or scattering volume of the troposphere.

These multipath signals have varying levels and relative phases, such that the aggregate received signal varies rapidly over a wide range of levels, and has significant phase dispersion about its median. Only a very small proportion of the signal energy passing through the common volume will be scattered, and only a small proportion of that will be directed towards the receiving station. Therefore the loss in the scattering process is extremely large and the angle through which the signal ray has to be deflected is an important characteristic of a troposcatter path; for best results it should be no more than a few degrees.

Troposcatter technology is mentioned in some texts as a 800 km technology characterized by intermittent communications at frequencies above 20 MHz, but preferably at VHF frequencies. Microwave transmission techniques developed in the early l950s, utilizing the troposcatter propagation mode, permit microwave transmission over path lengths that extend beyond the LOS. However, when compared with LOS, troposcatter is characterized by higher path loss, greater frequency fading and limited bandwidth. Figure 3-2 illustrates the difference between conventional multiple hop line-of-site radio links and direct troposcatter links between two distant "beyond the horizon" points.

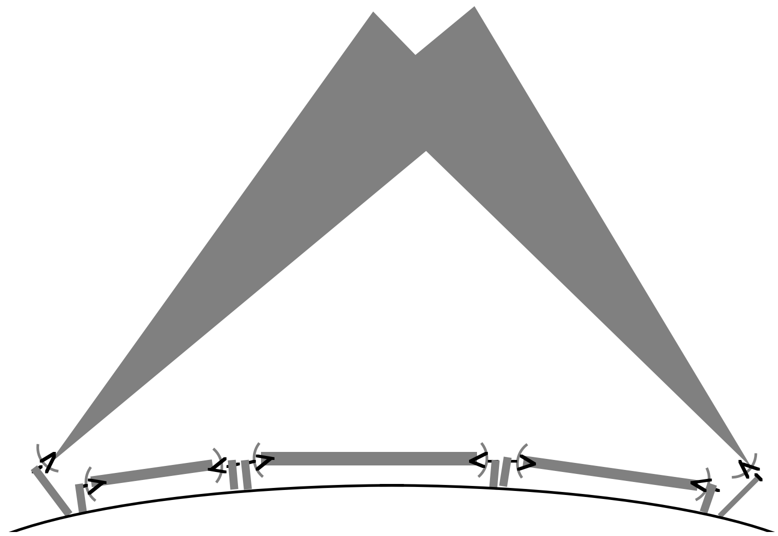


Figure 3-2. Line-of-Site/Troposcatter Communications

These disadvantages can be countered to a certain degree by the equipment design. In the 1960's many systems were installed to transverse long paths over difficult terrain. As examples, troposcatter became a principal means for long-haul multichannel systems for the Defense Communications System and for North Atlantic Treaty Organization (NATO). The systems introduced in the late 1950's and early 1960's, many of which are still in service, employed the same modulation and multiplexing techniques as the LOS microwave equipment of that time. Multiplex was Single Side Band-Frequency Division Multiplexing (SSB-FDM), and frequency modulation with pre-emphasis was used for the microwave equipment. With appropriate high power amplifiers, antenna systems, and diversity receivers, these systems provided traffic capacities of typically 24 to 60 channels and in some instances, 240 channels.

In recent years, digital techniques have been successfully applied to troposcatter. Because of technical and cost advantages, many civilian and military administrations are upgrading existing troposcatter systems and planning new construction on the basis of using digital techniques. A medium-size terminal communicates 250 km, at a 2 Mbps rate, with 1.85 kilowatt power at 4.4 to 5 GHz, 3.5 MHz bandwidth, Bit Error Rate (BER) of 10^{-5} and service time availability of 95%. It can accept up to sixty discrete 32 Kbps traffic channels; or one hundred twenty 16 Kbps channels with four 2 Kbps order wire circuits and a 16 Kbps voice order wire.

Troposcatter communications offer many advantages. These include:

- A signal-to-noise ratio characteristic which is relatively independent of propagation distance and characteristics. This offers better control of transmission quality
- Troposcatter offers a practical alternate to multi-hop microwave LOS transmission. In many applications, such as long over water paths where repeaters cannot be used, it is the only method of system implementation
- Where intervening terrain causes implementation problems, troposcatter can be the most cost effective technology
- Special modern techniques allow the digital troposcatter wider bandwidth to offer the unique opportunity of realizing intrinsic diversity from the multipath signal; and these modems are less sensitive to noise and interference

However, it possesses the following drawbacks:

- Digital troposcatter requires a relatively wide bandwidth. This is particularly true where 8-bit PCM, used for the Analog to Digital (A/D) process, requires 64 kHz per channel at baseband. However, even the more efficient continuous slope delta modulation at 32 or 16 Kbps, developed for the TRI-TAC and EUROCOM military systems, occupies greater RF bandwidth than analog systems of the same capacity
- Multipath dispersion of the propagation channels is the greatest technical problem for analog troposcatter. As a results, high reliability analog troposcatter systems typically are limited to 60 channels. (Advanced modems such as the Distortion Adaptive Receiver (DAR) not only make digital troposcatter practical but take advantage of the multipath to make it superior in all respects to analog troposcatter.)
- Available systems are expensive
- Antennas must be carefully aimed and have a clear view
- Because of the size and weight of troposcatter antennas and equipment, ample consideration must be given to installation. Portable systems require a heavy duty truck and trailer

For the Coast Guard, troposcatter transmission techniques can provide diversity for teleprinter services in some remote areas of the world for shore site to site communications. In the future we can expect to see the development of improved digital signal processing techniques further increasing the utility of digital troposcatter communications.

3.5 Meteor Burst

Meteor burst transmissions are propagated by VHF and UHF radio waves which either bounce off or are re-radiated from the ionization trails formed by small meteors entering the earth's atmosphere. The preferred transmission frequency range is 20 to 120 MHz (the lower the frequency, the better the performance). Maximum communications range of a single system is 2000 km; the optimum range is 650 to 1300 km; and ground wave propagation allows communications from 0 to 650 km. Communications range can be extended with additional master stations that operate in a store-and-forward mode.

Microprocessors provide packet formatting, buffering, error checking and control interaction between a master station and up to 1000 remote stations. Under ideal conditions digital data can be transmitted at a daily average of 100 words per minute in each direction in full duplex mode, or about 70 words per minute in half duplex mode. As a rule of thumb, for a system in half-duplex mode to transmit a 50 character message with a standard 1000-watt master station (using a 10-dB gain antenna) to a 300-watt remote station (using a dipole antenna) at a 4 Kbps data rate, the message waiting time will be less than two minutes at a probability of 0.90. Analog data, unless digitized, cannot be transmitted with modern, microprocessor-controlled systems because data is stored and then sent in short bursts as meteors become available. The present modulation method is Binary Phase Shift Keying (BPSK). Commercial frequency allocations allow a 20 kHz bandwidth, while Government allocations allow 25 kHz bandwidth. Meteor burst transmissions allow long range (i.e., up to 2000 km) continuous coverage with no time-of-day frequency changes and no channel interference and is inherently self-multiplexing and secure due to small signal footprint. However, it possesses inherent drawbacks, including:

- High susceptibility to noise (easily renders meteor burst technology useless in many real-world environments and requires that earth stations to be located at very quiet locations)
- Master earth stations require relatively high power and high gain/duplex antenna systems for adequate throughput and minimal communications wait
- Published throughput figures may only be realized in wilderness areas
- Cost amortized over small networks is high (typically $75K for master earth station installation, $7.5K for each remote station)

Its greatest utility to the Coast Guard may be for remote monitoring and communications applications where high throughput is not essential (e.g., would be low speed teleprinter service for Loran-C administrative and control circuits). These potential uses are currently restricted to terrestrial applications. Typical meteor burst radio equipment in use today use BSPK or Quadrature Phase Shift Keying (QPSK) modulation and readily available Yagi antennas (fixed frequency). A practical communications capability can be designed around **fixed** shore-based sites. The proper combination of the inversely related gain and beamwidth can be computed prior to deployment meteor burst communications gear to the **fixed** sites. However, implementation aboard mobile shipboard platforms today and in the foreseeable future is not feasible due to the non-availability of the requisite antenna and combiner (space diversity) technology.

With the trend of the rapid advancement of digital technology (microprocessors, error correction/modulation/diversity techniques), it can be anticipated that overall throughput performance and cost factors will improve substantially. If this indeed occurs, meteor burst communications technology can be used in place of many present day HF applications.

4 ANALOG (CONTINUOUS) MODULATION

4.1 Overview

Modulation is the technique whereby the characteristics of a carrier signal are varied in accordance with the changes in an informational signal. This section discusses continuous modulation techniques in common use today.

4.2 Narrow Band Frequency Modulation

Frequency Modulation (FM) is a process which makes the time derivative of a carrier's phase proportional to the baseband signal's amplitude. It is a form of angle modulation. The basic physical characteristics of an FM channel are as follows:

Modulation Index = Δ_n/n_m
Channel Bandwidth = $2\,(n_m + \Delta_n)$

Δ_n = Maximum frequency deviation
n_m = Highest information frequency

Since FM is considered to have an infinite number of side bands, a need for a narrow-band deviation developed. Narrow Band FM (NBFM) provides a middle ground modulation process between audio quality and spectrum conservation. Its common characteristics are as follows:

- Frequency deviation of plus or minus 5 kHz
- Bandwidth of approximately two times the message frequency
- Modulation index of one radian or less, and a single side band pair with significant amplitude

Its inherent advantages include:

- Enhanced spectrum conservation
- Relatively simple and inexpensive modulation equipment
- Ability for a wide band receiver to receive narrow-band signals, suffering minimal loss of audio in the detection process
- Broadcast reception interference can be reduced when used at frequencies below 30 MHz
- For voice communications, can be applied in most frequency bands (i.e., HF, VHF, UHF)

In spite of these advantages for voice communications, there exists an inherent tradeoff between the signal-to-noise ration and bandwidth – audio quality declines as the bandwidth is decreased. Thus, bandwidth must always be selected for the required degree of fidelity from the demodulation process. Nevertheless, the consideration of NBFM is most appropriate for Coast Guard systems requiring bandwidth conservation in either voice or teletype communications. NBFM will continue to play a significant role in future Coast Guard applications as this technology will continue to undergo steady improvement as it continues to be used in the areas of amateur radio, international broadcasting, Department of Defense (DoD) communications, and long distance aircraft and ship communications

4.3 Amplitude Modulation

Amplitude Modulation (AM) is a modulation scheme whereby the amplitude of a high frequency signal carrier is modulated by varying the signals at the rate and amplitude of the modulating signal as illustrated in Figure 4-1.

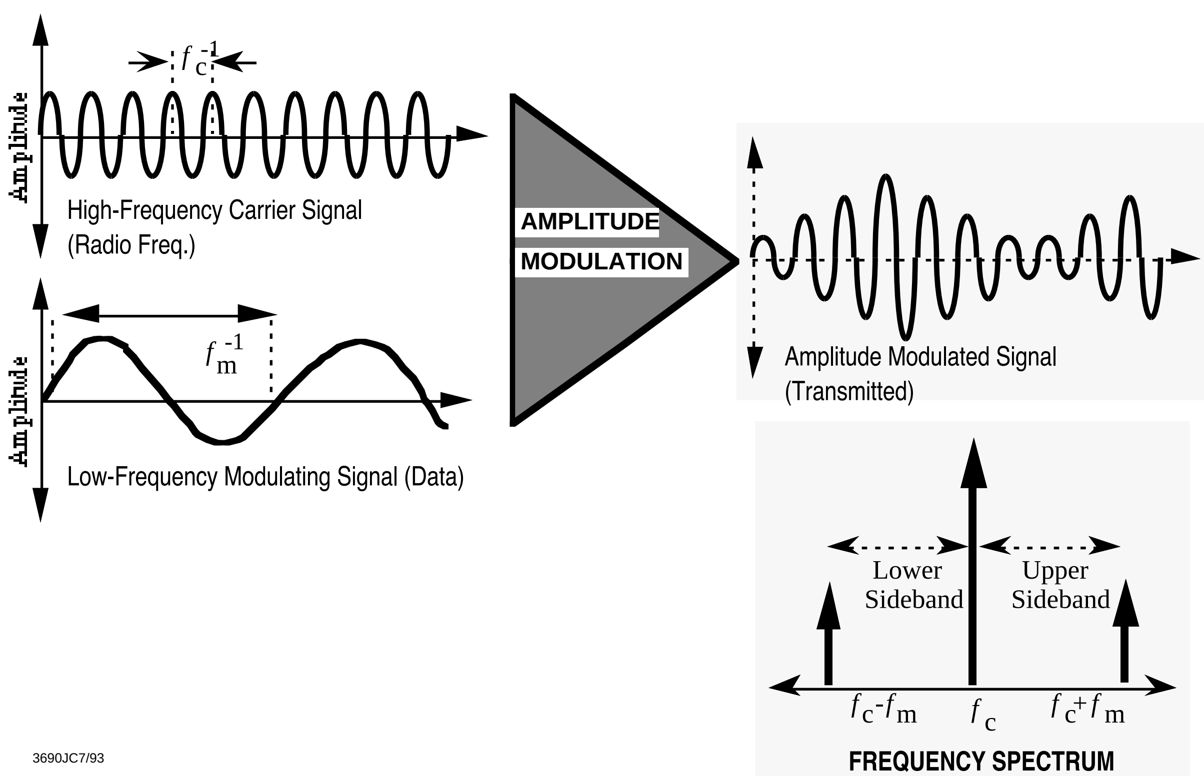


Figure 4-1. Amplitude Modulation.

When an AM signal is transmitted, the signal consists of the carrier and the Upper and Lower Side Bands (USB and LSB). The USB and LSB each contain all the information, although one is the inverse of the other. Therefore, normal AM is wasteful of both bandwidth and power because at optimum modulation levels, the carrier contains half the transmitted power, even though it contains no information. Single Side Band (SSB) is an amplitude modulation scheme whereby one of the sidebands and the carrier frequency are filtered out. SSB transmits only one side band and either the carrier or just the side band. Generally, the SSB process strips off the carrier and one of the side bands before final amplification so that output power is concentrated on carrying the information. To recover the information, the receiver offset (carrier) frequency must be exactly that of the transmitter. The SSB modulation technique typically requires one-fourth of the power used by conventional double side band AM to pass an equivalent amount of information. However, this technique is more complex and expensive than normal AM. Furthermore, speech has "sing-song" quality when the transmitting and receiving equipment are not precisely on-frequency.

Another technique, called Vestigial side band, provides a small amount of the carrier with the SSB (near side band suppression via gradual cut-off near carrier frequency). This permits the receiver to more easily match the transmitter frequency. Very little power is required for the vestigial carrier – just sufficient to provide to the receiver a frequency reference.

The use of AM by the Coast Guard is primarily voice traffic focused in the MF through UHF frequency bands. The MF frequency of 2182 kHz is monitored by most Coast Guard vessels and all Communications Stations (COMMSTAs) for distress and maritime safety voice transmissions. The VHF frequency of 121.5 MHz and 243.0 MHz are monitored by certain Coast Guard vessels and aircraft, as well as selected air stations for distress transmissions.

4.4 Independent Side Band

With SSB transmission, the carrier and one side band are removed from the signal. It is possible to replace the removed side band with another side band of information created by modulating a different input signal. Modulation may be accomplished with complete carrier suppression, or with any degree of carrier insertion desired. This is known as Independent Side Band (ISB) transmission. Both of the original signals have frequencies in the audio range, but in the transmitted signal each signal occupies a different group of frequencies in the spectrum. The process of distributing signals in the frequency spectrum so that they do not overlap is the process of frequency division multiplexing, or FDM. The key advantage in implementing ISB is its inherent ability to carry more information over an AM circuit utilizing the benefits of ISB, although the use of ISB will mean somewhat less power will be available per channel as compared with true SSB operation. The ability to rapidly squeeze as much information as possible over an HF channel is and will continue to be an area of great importance to the Coast Guard.

5 DIGITAL MODULATION

5.1 Overview

Pulse Code Modulation (PCM) is one of several Source Coding techniques available to a digital communications system. It is a process which converts an analog signal, representing message data, into a coded sequence of constant amplitude pulses in which the modulating signal wave form is sampled at regular intervals. However, PCM samples are first quantized into discrete steps; i.e., within a specified range of expected sample values, only certain discrete levels are allowed. All pulse modulation is based on the sampling principle which states that a continuous wave form that has a finite spectrum width which can be recovered from a set of discrete, instantaneous samples taken at a rate that is at least twice the highest signal frequency. For transmission, the PCM signals, which normally cannot be transmitted over long distances, in turn modulate a carrier by means of Frequency Shift Keying or other similar modulation scheme.

Although PCM is called a modulation technique, most modern texts address the subject as signal processing, related to sampled data for digital transmission. The output pulse sequence often is wired to adjacent digital processing circuits. PCM speech transmission systems providing excellent voice reproduction are widely used today. The various forms of PCM and interrelated techniques that are discussed in the ensuing paragraphs of this section [Differential Pulse Code Modulation/Adaptive Differential PCM, Pulse Width Modulation, Pulse Code Modulation, Log PCM, Delta Modulation, and Adaptive Delta Modulation (ADM)].

With PCM, analog data can be transformed into the digital domain where almost unlimited processing and transmission techniques abound. The advanced and modified forms of PCM can transmit data at lower data rates and more efficiently use the frequency spectrum. When spectral density varies widely with time, PCM can provide better performance for white (or nearly white) inputs. Compared to the analog signal that it represents, PCM requires a greater channel bandwidth. With the advent of Digital Signal Processing (DSP) chips and their continual increase in power, more powerful encoder equipment can be easily and economically built.

5.2 Log Pulse Code Modulation

A major problem with standard PCM is the very wide range of power levels which need to be handled. When quantizing, if the steps are uniform in size, the small amplitude signals will have a poorer Signal-to-Quantization-Noise Ratio (SQNR) than that associated with larger amplitude signals. The range within speech of one speaker, including the additional attenuation of connections before a PCM link is encountered, may result in a range of 60 dB. This 60 dB on a uniform step basis would involve approximately plus/minus 1000 levels. This impacts (increases) the cost and complexity of digital transmission systems based on standard PCM.

In order to alleviate this difficulty, companding was developed. In companding, the voltage of the wave form being transmitted is compressed. At the receiving end, the inverse operation occurs. The step sizes are tapered such that steps are closer together at low signal amplitudes and further apart at large amplitudes. In companding operations, low-level signals are lifted while the peaks of the high-level signals are maintained within a predetermined upper power limit. As a result, the low-level signal-to-noise ratio is significantly improved.

Signal compression, applied before quantization, is obtained by passing the signal through a non-linear network usually of the *A law* or *U law* type. The compressor's characteristics are such that at low amplitudes its slope is larger than at high amplitudes. Consequently a given signal change at low amplitude will carry the quantizer through many more steps than for the case at large amplitudes. The compression characteristic is one which gives constant gain to small signals changing to a logarithmic law after a particular signal level is obtained. Recovery of the distorted signal is obtained by using an expander at the receiver (hence the term "companding"). The major advantages gained from the use of this technique are:

- The ability to achieve uniform quality over entire ranges of PCM systems while simultaneously decreasing the required number of quantization steps
- The fact that companding greatly decreases the cost and complexity of the terminal conversion processes with digital transmission systems
- The inherent reduction in quantizing noise power that is obtained relative to a system with uniform quantization (typically in the neighborhood of 25 dB)

The only significant problem with this technique is the fact that as small amplitudes are increased, thereby improving the SQNR, a tradeoff must be considered in allowing a tolerable reduction in E_b/N for larger amplitudes. The use of companded PCM will continue to grow in step with advances in the technology of linear encoders, compressor/expander combinations, and signal compression techniques for video signals.

5.3 (Adaptive) Differential Pulse Code Modulation

Differential Pulse Code Modulation (DPCM) – also known as predictive coding and classified as wave form encoding – uses differences between samples, rather than their actual absolute amplitudes. For most data, the average amplitude variation from sample to sample is much less than the total amplitude variation. Therefore, fewer bits are needed to describe the difference. DPCM systems actually encode the difference between a current amplitude sample and a predicted amplitude value estimated from past samples. The estimation is made from the most recent input samples which are weighted. A similar algorithm decodes the difference.

Adaptive DPCM (ADPCM) is an adaptive predictive coding scheme using a sample-by-sample backward adaptive quantizer. In ADPCM, a short-time predictor with a relatively low order is ordinarily used for removing redundancy due to correlation amongst adjacent samples. The predictor is fixed for matching the long-term, average speech characteristics or is adaptively adjusted with the backward adaptation algorithm to match the time-varying characteristics of the input speech signal. The parameters of the predictor are optimized to obtain an efficient prediction in the sense that the mean square error between the predicted value and the true value of the signal is minimized. ADPCM further improves DPCM performance by increasing the number of predictor coefficients and dynamically adapts them to the signal. The only significant drawback to ADPCM stems from the fact that its performance degrades severely at channel bit error rates above 10^{-3}. Numerous implementations abound today. For instance, one widely used system employs a fixed predictor and an adaptive-step-size PCM quantizer with 64 possible step sizes. Differential Pulse Code Modulation schemes:

- Offers a practical method of coding signals efficiently without requiring large code book-memories
- Provide an estimated 10 dB SQNR improvement over PCM for speech, and 4 dB improvement for analog data transmission over telephone channels
- Implemented using sample sizes that can provide nearly the same voice quality as eight-bit PCM – this provides a 25% "bit saving" over standard PCM
- Implementing an adaptive scheme at an 8 kHz sampling rate can yield a 32 Kbps transmission rate and provide a SQNR of 30 dB over a 40 dB dynamic range
- Provide a better SQNR than Adaptive Delta Modulation (ADM) at transmission rates above 35 Kbps
- Are a proven technique for efficiently transmitting video images

Differential pulse code modulation is a well established technology in wide use today for applications involving the transmission of speech, video and data. Future trends in this technology area will undoubtedly include improved algorithms and quantizing techniques, coupled with advances in associated DSP chips. Future systems will undoubtedly cross-pollinate the various encoding techniques.

5.4 Pulse Width Modulation

Pulse Width Modulation (PWM), also known as Pulse Duration Modulation (PDM), or Pulse Length Modulation (PLM), is one of the major implementations of pulse time modulation. As with any technique employing pulse time modulation, the sampled values of an analog signal are encoded onto the digital signal's time axis. The message samples vary the duration or width of the individual pulses. Pulse amplitude is kept constant as the pulse width increases to a maximum at the analog signal's positive peak. The amplitude decreases to a minimum width when the input signal is at its most negative value. The modulating wave may vary the time of occurrence of the leading edge, the trailing edge, or both edges of the pulse. Additionally, PWM wave forms may be generated through either uniform or natural sampling. PWM provides great immunity to additive noise when compared to pulse amplitude modulation schemes. Further characteristics of PWM include:

- The baseband signal can be precisely recovered as the sample pulse is small compared to the time between successive samples
- The generation and recovery of PWM wave forms are fairly simple and inexpensive when compared to PCM
- Distortion can be minimized by increasing the sampling frequency
- The tradeoff between bandwidth and the signal-to-noise ratio of the modulated signal is much less severe for this wave form then for PCM
- The PWM is not as complex as other modulation systems such as PCM

Disadvantageous characteristics of PWM include the facts that:

- A wide bandwidth channel is required for its use
- Some distortion occurs when low-pass filtering is applied to the baseband signal (although acceptable limits do exist, on the average)
- Various levels of cross talk occur between baseband channels (although this decreases with increasing bandwidth)
- PWM is not as efficient with respect to power transmission as other modulation schemes
- PWM systems suffer a "threshold effect". As transmission bandwidth is increased, false pulses tend to occur more often. As a result, portions of the message data can be lost
- PWM spectra is quite difficult to evaluate due to the modulation's nonlinear characteristics

5.5 Delta Modulation

Delta Modulation (DM), also referred to as Linear Delta Modulation (LDM), is a simple predictive coding method that can be considered a special case of Differential Pulse Code Modulation in which the output quantizing level is two (one bit). Essentially, the output pulses produced by Delta Modulation reflect the derivative of the amplitude of the modulating input signal. Figure 5-1 illustrates how Delta Modulation samples a wave form.

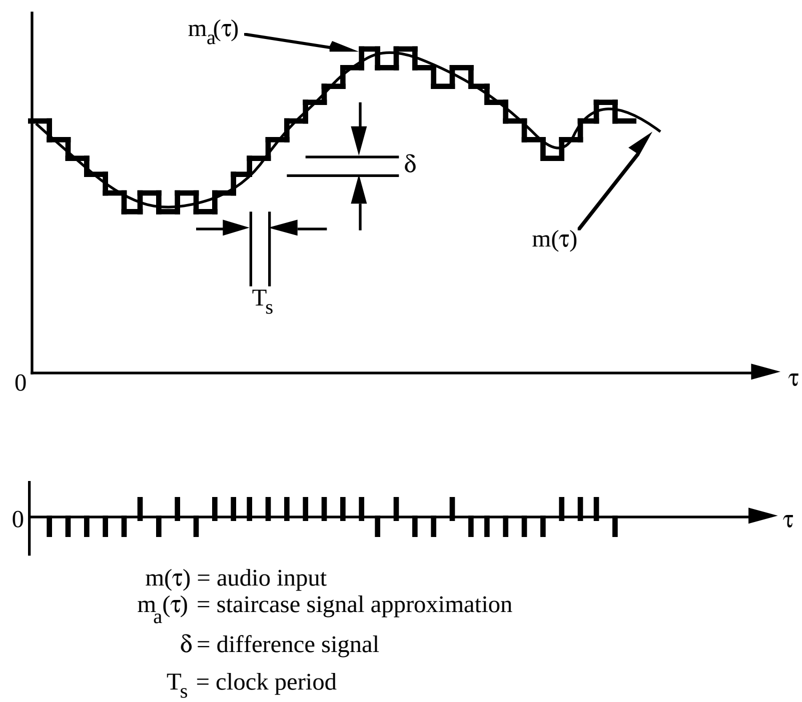


Figure 5-1. Delta Modulation – Sampled Wave Form

In a simple single-integration Delta Modulation, an encoder compares the audio input, m(t), with a predicted audio value that is derived from previous samples. The encoder's modulator produces positive pulses when the difference signal d is positive. It produces negative pulses when the difference is negative. The resulting output pulse train is then integrated and compared with the next input signal, thus closing the encoder feedback loop. In this manner, the input signal is approximated by increments at each clock period, T_S. At the receiver the pulse train is integrated and filtered to produce a delayed approximation to the original signal. Delta Modulation can be implemented very simply (output is an uncomplicated serial pulse stream). Its signal-to-noise ratio at a 40 Kbps sampling rate is equivalent to that of a five-bit PCM system. For Delta Modulation systems, the analog input must be frequency and amplitude limited. Delta Modulation can accommodate signals with only limited dynamic range due to noise caused by slope overload and granularity. Slope overload occurs when the quantizing step size is too small to follow steep segments of the input wave form. Granular noise occurs when the quantizing stair case hunts around a relatively flat segment of the input wave form, with a step size that is too large relative to the local slope characteristics of the input. Thus, Delta Modulation is particularly well suited for speech signal encoding. This is due to the fact that a speech signal reaches a maximum slope at a relatively low frequency around 800 Hz for vocal utterances (the spectral density of speech falls off at 6 dB per octave above 800 Hz – even more rapidly above 1600 Hz).

5.6 Adaptive Delta Modulation

Similar to delta modulation, Adaptive Delta Modulation is a predictive coding method which is a special case of Differential Pulse Code Modulation. In Delta Modulation, the output quantization level is two (one bit). Adaptive Delta Modulation differs from this as the quantizing step size is variable and/or "adaptive" to the input signal. As a result, slope overload and granular noise are eliminated. Adaptive Delta Modulation systems are superior to Differential/Adaptive Differential Pulse Code Modulation systems at transmission rates less than 32 Kbps for speech and voice band data applications.

The various Adaptive Delta Modulation schemes differ in quantization, prediction, and adaptation strategy. Some of the common implementations are summarized below:

- <u>Continuous Variable Slope Delta (CVSD) Modulator.</u> This adaptive technique exploits the syllabic characteristics of speech wave forms to continuously vary quantizing step size and minimizing the number of bits required in describing the speech. A comparator generates an error signal between the band-limited input speech and the prediction filter output. A one-bit quantizer samples the error signal in accordance with the desired transmission rate. The quantizer step size is controlled by a syllabic compander which consists of a three or four-bit shift register, a comparator, and a simple RC low-pass filter. Algorithms call for step size enlargement when the comparator detects three or four consecutive binary l's. Conversely, step size decrease when there are alternating 1's and 0's
- <u>Song Mode Voice Digital Adaptive Delta Modulator (SVADM)</u>. At each sampling point, the algorithm produces a quantizing step size which minimizes the mean-square error between the encoder input signal and the final processed speech at the decoder output. It has word intelligibility of 99% at 16 Kbps and 90% at 9.6 Kbps. It also offers a 40 dB dynamic range. Signals having amplitudes ranging

from ±50mV to ±5Y can be encoded with approximately the same signal to noise ratio

- Modified ABATE Algorithm. This scheme is adaptively capable of following the received signal on an extremely noisy channel (error rate of order 10^{-1}). The average number of erroneous step sizes following a received error is less than other popular Adaptive Delta Modulation algorithms. It maintains high word intelligibility and reasonable voice quality at sampling rates of 24 Kbps or 32 Kbps in the presence of very high channel errors. NASA chose this algorithm for the space shuttle
- Jayant's Adaptive Delta Modulator. This technique is commonly called Constant Factor Delta Modulation (CFDM). CFDM uses instantaneous, exponential adaptation in step size companding. That is, the quantizer step size is changed at each sampling time by one of two specific factors, according to the states of the present and previous bits. Speech limited to a 3.3 kHz bandwidth and sampled at 60 kHz (with a Jayant coefficient of 1.5) showed a 10 dB increase in output signal to noise ratio compared to conventional DM
- Hybrid Companding Delta Modulation (HCDM). This method uses both syllabic and instantaneous companding schemes. The quantizer basic step size is determined by the coded signal slope energy that is estimated every five milliseconds

In summary, adaptive delta modulation schemes can be summarized as follows:

- They provide a great deal of flexibility as far as providing effective means for performing tradeoffs between quality (signal) and robustness (system)
- Due to feedback quantizers, adaptive delta modulation can significantly outperform PCM quantizers for input signals for which the power spectral density is non-white (e.g., speech, video)
- The required bandwidth is relatively narrow
- The Continuous Variable Slope Delta Modulator systems can be made relatively insensitive to speech errors (with a corresponding degradation in quality in the reconstructed speech). High quality speech coding has been demonstrated with Continuous Variable Slope Delta Modulator systems at bit rates less than 25 Kbps
- The Song Mode Voice Digital Adaptive Delta Modulator algorithm is easily implemented and produces high quality speech at fairly low bit rates. Unlike the Continuous Variable Slope Delta Modulator scheme, which is specifically designed for encoding speech, the Song Mode Voice Digital Adaptive Delta Modulator is

also applicable to non-speech signals. It has wide (40 dB) dynamic range

- The Modified ABATE algorithm provides high performance in the presence of channel errors.
- The Hybrid Companding Delta Modulation is superior to other adaptive and differential encoding schemes at low bit rates.
- The Adaptive Delta Modulation scheme implemented at less than 32 Kbps is the algorithm of choice (preferred over Adaptive Differential Pulse Code Modulation) for speech and voice band data applications due to its overall performance and encoder simplicity. The "threshold" between Hybrid Companding Delta Modulation and Adaptive Differential Pulse Code Modulation systems occurs in the neighborhood of 35 Kbps bit rate.

These delta modulation schemes will continue to be highly applicable to most voice and data digital transmission Coast Guard applications. Furthermore, with the continuing advances with Very Large Scale Integration (VLSI) technology, it can be anticipated that intense development in voice and data coding will continue. Thus, we can anticipate more economical and higher performance systems employing more sophisticated algorithms.

6 RADIO FREQUENCY MODULATION

6.1 Phase Shift Keying

PSK is a band pass signaling technique to modulate binary data (e.g., binary PCM from digitized voice, digital computer data) onto a RF carrier. In PSK, the amplitude and frequency of the carrier stay constant while the carrier phase is switched between two or more values. PSK is commonly used on channels which are power-limited rather than band limited. The transmitted power spectrum contains significant energy in a bandwidth equal to approximately twice the data rate.

With binary phase-shift keying, also known as BPSK, the carrier phase has one of two constant values representing a zero or a one. An additional extension of BPSK is the well known (and well used) QPSK multilevel signaling technique.

The QPSK signal has the same bit error performance as BPSK, but the bandwidth occupancy is half as great. QPSK gives the best overall performance in terms of the minimum bandwidth required for a given signaling rate and one of the smallest probability of errors for a given E_b/Noise. It is generally used to conserve bandwidth. The bandwidth of the QPSK signal will be exactly one-half that of BPSK signal for a given bit rate. Unfortunately, QPSK is relatively expensive to implement since it requires coherent detection.

Band pass digital signaling systems used by the Coast Guard could possibly employ this signaling technique if bandwidth conservation is of prime concern for a given probability of error. Narrow band QPSK channels using an FDM format within an HF SSB channel for collecting radio navigation data is a strong implementation candidate for QPSK.

With the advent and evolution of cellular communications and faster processor chips to implement algorithms, PSK variations are being studied with the hope that multiple channel interference and thermal noise can be minimized economically. For instance, recent studies have shown that Differentially Encoded Phase-Shift Keying (DPSK) signal transmission with differentially coherent demodulation is an appropriate choice when circuit simplicity and the accompanying cost effectiveness are considerations on a land mobile radio channel.

6.2 Minimum Shift Keying

Minimum Shift Keying (MSK) is a spectrally efficient modulation scheme which has been developed to alleviate the spectral congestion caused by increasing demand for digital RF channels. MSK conserves bandwidth, yet possesses a prescribed average bit error rate with minimum signal power. Other desirable properties include a constant envelope and associated simple demodulation and synchronization circuitry.

Briefly, MSK can be viewed as a special case of the Continuous Phase Frequency Shift Keying (CPFSK) with a frequency deviation equal to half of the bit rate. This frequency deviation is the minimum frequency spacing which allows the two FSK signals to be coherently orthogonal. MSK is sometimes viewed as a form of Offset Quadrature Phase Shift Keying (OQPSK) in which the symbol pulse is a half-cycle sinusoid rather than the usual rectangular form. Whether MSK is observed as a form of FSK or OQPSK, it combines many desirable attributes into one modulation scheme. MSK is used in many systems, including military tactical radio, ELF underwater communications, and domestic (e.g., AT&T) satellite systems. It possesses the following characteristics:

- With its smoother pulse, it possesses lower side lobes than QPSK or OQPSK thereby providing a more compact modulation scheme
- Without phase changes, MSK greatly suppresses out-of-band interference with other communications systems
- Demodulation and synchronization circuits required are simple
- It possesses the same bit error rates as BPSK, QPSK, and OQPSK
- It is used in both Time-Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA)
- It is an excellent modulation technique for digital links when bandwidth conservation and efficient amplitude-saturating transmitters are significant requirements
- It is not preferred for narrow-band satellite links due to its wide main lobe

6.3 Amplitude Shift Keying

When transmitting binary data (i.e., binary PCM waves obtained by digitizing voice or video signals, digital computer data), it is necessary to modulate a carrier wave using some type of generalized modulation technique. A common binary band pass signaling technique is known as Amplitude Shift Keying (ASK). It is a well-established modulation technique in use today. In fact, it was one of the first modulation techniques used. ASK involves the keying (switching) a carrier sinusoid on and off. A binary "1" is represented by transmitting a sinusoidal carrier wave of fixed amplitude and fixed frequency for the bit duration t seconds, whereas binary "0" is represented by switching off the carrier for t seconds. This modulation technique is also known as On-Off Keying (OOK). The significant advantage in employing this scheme stems from the fact the associated demodulation circuits are simple and inexpensive. Synchronous (matched filter) or envelope detection circuitry can be employed. Nevertheless, ASK does possess the following drawbacks:

- Its use for in radio channels is hindered due to the difficulty in maintaining appropriate amplitude levels (more commonly employed in wire transmission circuits where accurate control over signal amplitudes can be maintained)
- For a given bit error rate, the energy required to maintain the rate increases with the number of signals squared

Presently, other modulation schemes, such as FSK or PSK are the choice means for transmitting binary data over RF. There are no inherent advantages in pursuing the utilization of this scheme for Coast Guard RF transmission applications.

6.4 Frequency Shift Keying

FSK is a method of transmitting digital information using a sequence of carrier pulses of constant amplitude and differing frequencies. FSK covers a variety of keying methods, but its purest form is called coherent FSK. This is frequency modulation in which the modulating wave shifts the output frequency between two predetermined values without phase discontinuities. Non-coherent FSK, or Frequency Exchange Keying (FEK), is simpler and usually is created by alternately keying one of two independent crystal-controlled oscillators at the source. This is sometimes referred to as two channel on-off keying.

Another common form of FSK is one in which the source shifts the frequency of a small audio oscillator (coherent FSK), which is then used to modulate the transmitted wave using either FM (less commonly AM). This technique is more precisely called Audio Frequency Shift Keying (AFSK). AFSK sent FM USB is indecipherable from coherent FSK except for the frequency offset of the side band (AFSK sent LSB would be inverted). FSK and AFSK are used for short-haul wire transmission of telephone supervisory and numeric signals, telegraphy, teleprinter, and facsimile transmissions.

Lower probabilities of error are achieved with coherent detection than with non-coherent or envelope detection, and by using a binary shift rather than more frequencies. As the Bit Error Rate (BER) decreases, there is a corresponding decreasing advantage to use coherent over non-coherent FSK. Therefore, at very low BER, FEK is roughly comparable in performance and very much simpler to implement. Coherent FSK performance is approximately 3 dB less than PSK at comparable BER. Performance is not degraded, however, when going from FSK to Multiple FSK (MFSK) as is the case when going from PSK to QPSK. Therefore, FSK is simpler and offers comparable or better performance than PSK at higher levels of multiplexing. Systems attempting to increase speed with the same bandwidth over RF have gotten away from FSK.

Another technique, the Duo-binary, offers increased speed without an increased bandwidth with only a small sacrifice in noise immunity. Duo-binary uses a three-level signal which may only change one step at a time. The resultant signal can be used to modulate an audio tone by either AM or FM.

7 INFORMATION CODING/PROCESSING

7.1 Channel Coding

Channel coding/encoding refers to the data transformation, performed after source encoding but prior to modulation, that transforms source bits into channel bits. There are two types of channel encoding; wave form (or Signal Design) coding and Structural Sequence coding.

Wave form coding is any source data transformation that renders the detection process less subject to errors and thereby improves transmission performance. It increases performance in an overall sense because the encoding process produces a signal set with better distance properties than the original. Wave form coding signal sets used most extensively include the following:

- M-ary signaling-orthogonal
- Bi-orthogonal
- Trans-orthogonal (simplex) signaling

Structured sequence encoding, by comparison, improves performance by embedding the data with structured redundancy, which then may be used for transmission error detection and correction. Structured sequences are partitioned into two important subcategories: block coding and convolutional coding.

With block coding, the source data is first segmented into blocks of q data bits each. Each block can represent any one of $M = 2^q$ distinct messages. The encoder transforms each message block into a larger block of n digits. This set of **M** coded messages is called a code block. The (n-q) digits, which are added to each message block, are called redundant digits. They carry no new information. Realizing that all code words have an equal likelihood of being transmitted, an optimum demodulation scheme is used. Conditional probabilities are calculated for all possible code words received and compared. The maximum conditional probability indicates the correct code word and hence limits any transmission errors.

Convolutional coding requires more substantial detail to describe adequately and goes beyond the scope of this document. Knowledge of shift registers, modulo-2 sumers, state diagrams, transfer functions, and various decoders (sequential or Viterbi) is required. For further information refer to *A Structured Overview of Digital Communications – A Tutorial Review-Part II*.

Channel coding improves the effective bit-error-rate performance of a channel at the expense of bandwidth and additional power requirement. When selecting a channel coding scheme, power and bandwidth considerations must be traded-off against error performance. New channel coding schemes are being developed at a rapid rate as continuing research is being performed in the areas of sub-class of block codes known as cyclic codes (note: cyclic codes are easily implemented and their decoding methods are simple and efficient), block distortion-analysis, minimization, and measurement.

7.2 Error Detecting and Correcting Techniques

Error detecting/correcting may be carried out at several levels of sophistication. Within certain limits, errors can be detected with the transmitted signal at the receiver and corrected. To accomplish this, symbols must be added to the transmitted information as a "check", and the check must relate to a finite section of the transmission. Hence, systems divide the incoming signals into blocks or words, and check each word/block independently. Two approaches can then be used. First, checks can be appended to each word, such that the receiving equipment can deduce the presence and the location of an error directly. As an alternative, the checks may only reveal that an error has been introduced, and to correct this the receiving terminal then requests a re-transmission of the block in question.

Even if re-transmission is not requested, the error detecting system remains. This system is adequate if the transmission is such that a certain error content is acceptable, and the receive site only needs to know when the error rate is becoming excessive. Alternatively, it is possible to conceal errors without actually correcting them. An incorrect sample can then be replaced by an estimate of its probable value, which in most cases will be either the last correctly transmitted value, or an interpolation between the preceding and following error free samples.

Correction of this type still requires the detection of every error, but cases also arise where even this is not needed, and it suffices to obtain an indication of the frequency of errors. Usage of error monitoring schemes to detect the presence of an error condition are sometimes preferred over the comprehensive error detection scheme, and therefore can be introduced at a large number of locations in the system. As a result, the point at which the error condition occurred can be identified.

An adequate explanation of specific error detection/correction techniques is beyond the scope of this document. The following briefly describes some of the common techniques in use today:

- <u>Parity Check</u>. Parity checking is considered the simplest method of error detection. An additional time slot is added to give odd or even parity indications. This scheme will clearly detect all single errors, or any combination of an odd number of errors in a word, but will fail to detect any even number of errors. It will not indicate the number of errors that have occurred. Hence, it is unsuitable for use in cases where high error rates occur, or where errors may appear in bursts.

- Hamming Code. (With this scheme, m information digits are associated with k *parity*-check digits to form a word length $n = m + k$. Each parity check is associated with a certain group of digits within the word, such that if any single error occurred, the resultant pattern of parity-check violations indicates its position within the word, thereby providing a single-error correcting code. Position of the error is given with a binary representation. The hamming technique used today is the (7,4) code. It is a code with 7-digit words containing four information digits. Additionally, further modifications are being made to the hamming code on an experimental basis.
- Spectrum Shaping. Spectrum shaping is used in advance to minimize interference (cross talk) between different channels. This is almost invariably caused by the high-frequency end of the signal spectrum. The total interference power can be significantly reduced by minimizing the high-frequency content of the transmitted signals.
- Error Correcting Codes. The following are some of the more significant transmission codes that have been developed for digital transmissions
  - *Binary Codes*. These are formed by taking a block of n message digits and converting it into a new block of m digits for transmission (m>n). The message digits are not changed, and the re-coding is carried out by systematic logical operations.
  - *Bipolar Codes* (alternate mark inversion). In this coding process alternate positive and negative pulses are used to represent marks. There is a built-in error detecting capability, since errors will introduce alternate sign violations of the mark. Redundancy is considered to be in the neighborhood of 60%.
  - *4B3T Codes*. This scheme provides a more effective use of the available transmission capacity. Conversion of binary digits to ternary digits is first performed. Careful pairing of binary and ternary groups using mode alternation techniques follows next. Redundancy is reduced to 20%.

Much ongoing research and development efforts have been under way in the field of error detection and correction. Specifically, much work is being performed the area of error codes (e.g., linear block codes, cyclic codes, codes based on spectral techniques, convolutional codes). Likewise, much complementary work is proceeding in the silicon implementations of error detection/correction technology. Inevitably, trend towards higher performance and cheaper "off-the-shelf" solutions can be expected to continue.

7.3 Digital Signal Processing

Digital signal processing is the manipulation of digital information by digital processors. Analog information often is converted to the digital domain for processing by either large computers or small, dedicated microchip electronics. Digital processing can replace or improve any operation that can be done with the traditional analog tools; resistors, inductors, capacitors (RIC) circuits, mixers, limiters, and rectifiers.

The major subdivisions of digital signal processing are digital filtering and spectrum analysis. Digital filtering is further divided into Finite Impulse Response (FIR) filters and Infinite Impulse Response (IIR) filters. Spectrum analysis is broken into calculation of spectra by the Discrete Fourier Transform (DFT) and by statistical techniques.

For many applications, digital filters provide significant advantages over analog filters. These include:

- No short-term drift with temperature and/or minor supply voltage fluctuations
- No long-term drift with age
- Suitability for multiplexing
- Accuracy based on the algorithm and/or the number of bits
- Ideally suited to process data available in binary form
- Near identical performance of prototypes and final products as compared to analog products which often need individual "tuning"
- Predictable results – ability to perform simulations on a workstation or mainframe
- Ability to change prototypes simply via software changes (i.e., software processing a 4th order elliptic filter can be changed to the 5th order via code modification much easier than a similar analog filter could be changed)
- Superior performance in many applications (e.g., filters with no phase distortion, true parallel bank of filters, no-drift modulators)
- Unlimited performance capabilities no analog system could hope to achieve (e.g., spectral estimation, satellite trajectory model fitting, non-linear parameter estimation, arbitrary-order filtering, non-casual smoothing)

There are, however, some significant disadvantages associated with digital filters. These include:

- Relatively high per-unit cost (although prices are dropping rapidly)
- Inability to process huge bandwidth signals (100 MHz) due to limitation of digital hardware (processing speed) and the large amount of clocking and synchronization circuitry required

Digital Signal Processing will play a key role for the Coast Guard. As the high-tech arena expands almost all electronic equipment purchased or developed by the Coast Guard (other than radio frequency and carrier subsystems) will operate, internally and externally, within the digital domain. Coast Guard efforts in the C2 Test Bed and ARQ systems evaluation use digital communications technology although the phrase, "Digital Signal Processing" technically is reserved for digital filtering or spectrum analysis.

The integrated circuits, themselves, are becoming more sophisticated. Each year, the number of components per chip doubles.

The current trend in the area of digital signal processing has been one of rapid advances in the technology and its application. Digital circuits are becoming more highly integrated and significantly cheaper. This is a trend which will undoubtedly continue into the foreseeable future. Advancing VLSI technology is bringing to market both higher density memory chips as well as extremely fast processing capabilities. Technology from the DoD Very High Speed Integrated Circuits (VHSIC) program has produced digital technology to perform high throughput signal processing which was digitally impossible just a few short years ago. As the Digital Signal Processing technology evolves, we can expect faster processing power through parallel processors (e.g., neural networks) to reduce or eliminate the bandwidth constraint.

7.4 Data Compression

Data compression is the technology that minimizes data quantity without sacrificing information. In a digital communications system, it is part of the source coding process. Many authors consider that source encoding includes both digital formatting and data compression.

Two software techniques that can result in a more effective encoded data representation are logical and physical data compression. Logical compression, while being data dependent, eliminates redundant information fields and represents the remaining elements with as few logical indicators as is feasible. For instance, if a data field contains 20 alphanumeric positions for accommodating all possible entries, many positions may waste space by being blank. A numerical or binary representation of the entries would be much more efficient.

Physical compression reduces data quantity prior to transmission, and expands the data to its original format upon receipt at a distant location. Unlike logical compression, physical compression takes into account a character or group's frequency of occurrence. One expert, Gilbert Held, cites nine physical compression techniques. They are:

- Null Suppression. Repeated nulls or blanks are substituted by an indicator and count.
- Bit Mapping. Binary bits keep record of repeated characters.
- Run-length Encoding. Similar to null suppression but for repeat characters as well.
- Half-byte Packing. Two numeric values can be packed into one character if enough adjacent bits are the same.
- Diatomic Encoding. Character pairs are replaced by a special character.
- Pattern Substitution. A special character code is substituted for a pre-defined character pattern; it is a sophisticated form of diatomic encoding.
- Relative Encoding. Original data run streams that vary slightly are broken into patterns relative to each other.
- Forms Mode Operation. Less fixed data is needed since standard form data is known.
- Statistical Encoding. Short codes are used for frequently occurring characters or groups, while longer codes are used for frequently occurring ones.

Techniques which are hardware-related, yet dependent upon compression algorithms, include:

- Differential Pulse Code Modulation (DPCM). Differences between current and predicted amplitudes are encoded.
- Delta Modulation (DM). A special case of DPCM where the output quantizing level is taken to be one bit.
- Adaptive Delta Modulation (ADM). Where a system adaptively varies the gain over a continuous range.
- Linear Predictive Coding (LPC). Which encodes significant process features rather than encode wave form samples.

Certainly new and more sophisticated compression methods will continue to proliferate (e.g., commercial use of fractal compression). Figure 7-1 roughly depicts the current (circa 1993) state of compression capabilities.

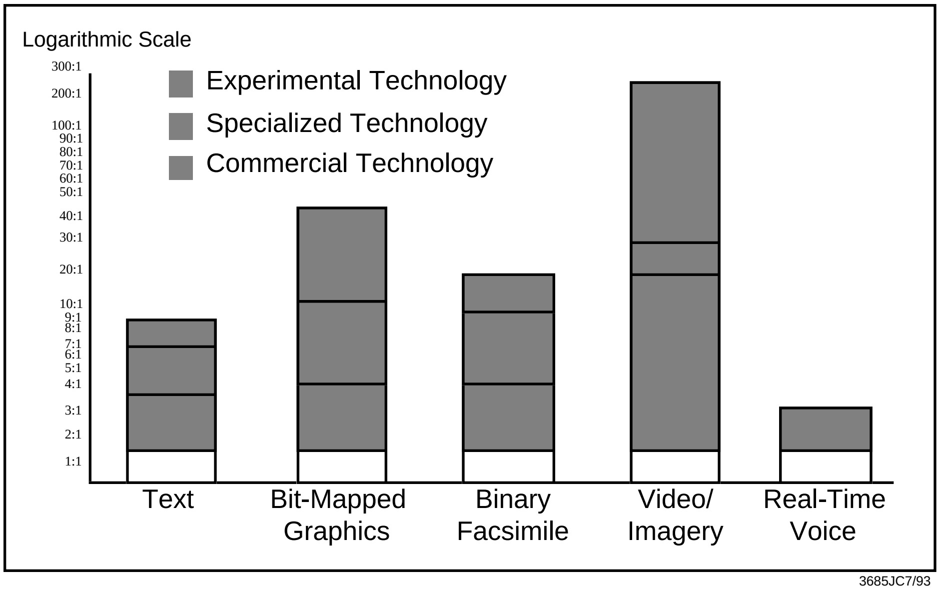


Figure 7-1. Achievable Compression Ratios (c. 1993)

7.5 (Adaptive) Transform Coding

Transform or block coding is a data compression technique by which a set of source elements is coded as a unit. The term "transform" indicates that the original set of elements is processed by an invertible mathematical transformation prior to encoding. Figure 7-2 depicts the block diagram representation of a transform coding/decoding system.

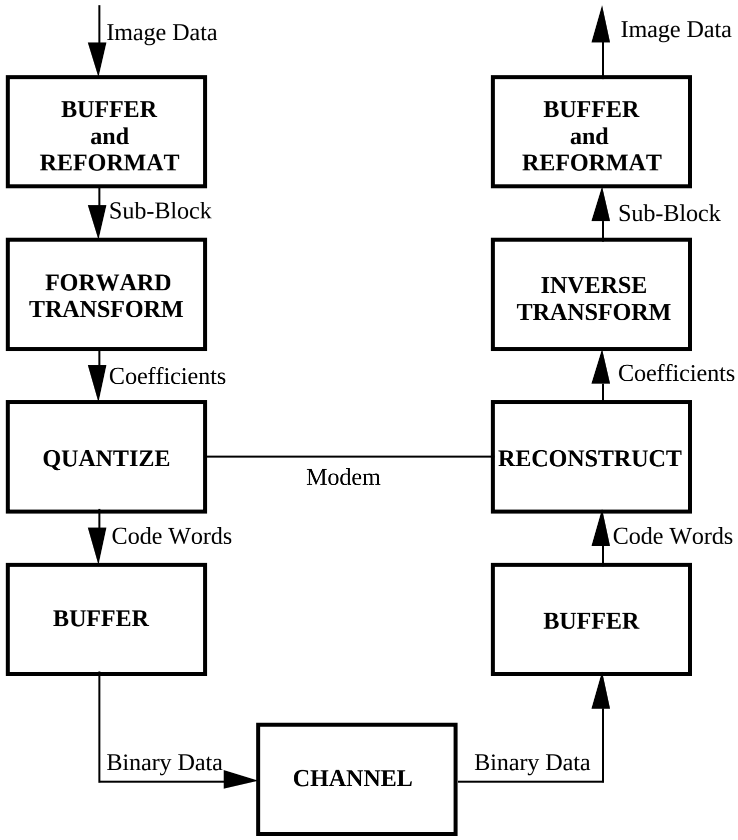


Figure 7-2. Basic Transform Coding System

The coding unit consists of a reformatting memory followed by the transformation, and finally, the actual coding process. The coding process, considered to occur in two steps, consists of a de-correlation process followed by the application of a memory-less quantizer.

Quantization, a primary consideration for all source coding procedures, is the mapping from the analog source to its quantized equivalent. Distortion due to quantizing is expected, but efforts to minimize it must be taken.

Consequently, a means to minimize quantizing noise led to the development of Adaptive Transform Coding. Figure 7-3 illustrates the adaptive control loop which has been added.

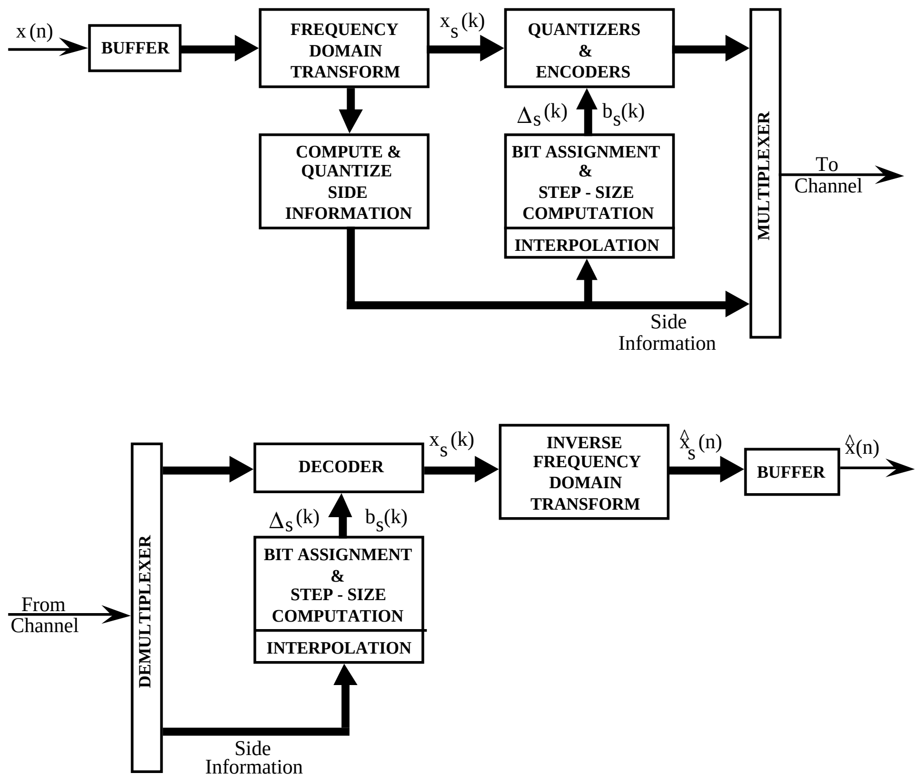


Figure 7-3. Block Diagram – Adaptive Transform Coder

The side information is spectral prediction information obtained at the transmitter and transmitted to the receiver. This information, in short, controls the step size of the data quantizer and takes into account overload and granular noise within the coder. Some key points regarding adaptive transform coders are:

- Unlike vocoders, the encoding process for wave form coders (adaptive) is not totally dependent upon speech production and speech perception mechanisms
- Frequency domain coders can match and exceed the quality of their time domain counterparts – mechanisms of speech production and auditory masking can be conveniently modeled in the frequency domain
- Adaptive techniques significantly outperform non-adaptive procedures.
- Implementation of adaptive techniques require a significant degree of added complexity as compared to non-adaptive procedures

Adaptive Transform Coding techniques for speech communications have been receiving considerable attention from researchers and users. This trend should continue. We can expect the following trends with respect to this technology:

- Improved frequency domain transforms (transform algorithms), particularly in the areas of computability and size
- Advanced buffering concepts, quantization, and block overlapping, bit allocation, and adaptive control (utilization of side information)

7.6 Linear Predictive Coding

Linear Predictive Coding (LPC) is one of the most important processing techniques for narrow-band digital transmission of speech. Developed in 1968, LPC is classified as "parametric encoding" which uses a parametric representation of the speech signal to reduce the necessary bit rate. (Note that the other major classification of speech coding systems is "wave form coders" that strive for facsimile reproduction of the signal wave form). At a receiver the parameters are synthesized into a replica of the original voice. Source coders are often called, vocoders (voice coders). LPC technology produces relatively good quality speech when used at rates between 1200 and 4800 bps.

In general, algorithms model human speech with the classic acoustic tube, excited by a sound source. The tube is the talker's vocal tract which actually changes shape, and thus the frequency spectrum with every sound. Although it changes shape from one sound to the next, it can be modeled as approximately stationary for any 20 millisecond period. The model is the heart of the LPC because it infers, with no mathematical powers higher than one, that each new speech sample is predictable from the previous sample. Usually the excitation model is either a train of periodic pulses (corresponding to voiced speech, e.g., vowel sounds) or random noise (corresponding to unvoiced speech, e.g., consonant sounds).

An analyzer section of a conventional LPC determines, during each 20 milliseconds, eight to ten model coefficients, and determines pitch, fundamental frequency, voice/non-voice for selecting pitch or noise excitation, and an amplitude coefficient. These parameters are then used to synthesize the original speech.

LPC systems, as well as other speech processing systems, can be built economically and easily with DSP chips. They offer fairly good quality speech reproduction at low bit rates – 4800 bits/second is usually the upper limit. When using the Vector Quantization technique, 800 bits/second can produce slightly degraded speech quality. A 150 bits/second rate can produce intelligible speech. LPC is one of the leaders in terms of the "naturalness" of voice that can be achieved with a given level of compression. LPC, being the basis for most synthetic voice production, e.g., computer voices or artificial speech, has opened a new chapter in intelligent human-machine interaction. Almost all feedback to humans can be implemented by synthetic voice. However, synthetic speech as produced with LPC is not completely natural sounding.

Many flaws still persist in the reproduced speech. As the coding efficiency goes up (and the bit rate goes down) the speech quality deteriorates.

LPC is a major contender in the explosive development of speech processing systems. Many versions of LPC have been and are being actively developed. For more efficient coding there already exists advanced techniques such as Vector Quantization, Multi-Phase Analysis, Simplified LPC, and Line Spectral Pairs. Given the rapid advance of VLSI technology, it seems likely that increasingly complicated and efficient algorithms will be implemented and rapidly brought to the marketplace.

7.7 Formant Vocoder

Formant voice coders (vocoders) are classified as "parametric encoders" since parameters represent the speech signal. At a receiver the parameters are synthesized into a replica of the original voice. This technology does not produce high quality speech, but allows artificial speech to be implemented at very low bit rate and with a minimal storage requirement.

"Formant" is the name given to the spectral energy bands in human speech. The centroids of the energy bands trace out trajectories which constitute the modulated information responsible for perception of the various speech utterances. The analyzer portion of a formant vocoder attempts to dynamically track the formant centroids in the original speech signal message. These vocoders require a lower data rate than LPC systems because there are fewer parameters. They are especially suited for artificial speech systems as the speech is "natural" sounding. Early formant coders represented a message with eight parameters – three formants, pitch frequency, the voice/non-voice "decision" parameter, and individual formant amplitudes. Some second generation formant coders use 16 parameters; 7 of them are independent and 9 of them are dependent variables. These parameters are then used to synthesize the original speech by means of a unique architecture. Low bit-rate, formant-based speech synthesizers are ideal for low-power, monolithic Complementary Metal Diode Semiconductor (CMOS) implementation, using either digital or switched-capacitor filters. Second generation formant speech synthesizers yield speech of higher quality than low-end LPC systems at average bit rates of 500-600 bits/second, and perform better than common wave form compression methods for many speakers. These coders can operate over a bit range of 200 to 2000 bits/second with speech quality directly proportional to the data rate. Formant vocoders are memory-efficient. Large vocabulary synthesizers are possible because of the lower bit rate and fewer parameters. Yet, these synthesizers contain all functional characteristics necessary to reproduce every principle phonetic component of spoken English – vowels, voiced and unvoiced fricatives, nasals, stops, voice bars, aspirates and pauses. Implementation specific switching and filter architecture eliminate redundancy and optimize the implementation for monolithic applications.

Formant vocoders will be among the contenders in the proliferation of speech processing development that has begun over the past five years. It seems likely that increasingly complicated and efficient algorithms will be developed with the advance of VLSI technology.

7.8 Simplex Teleprinting Over Radio

Simplex Teleprinting Over Radio (SITOR), Teleprinting Over Radio (TOR), and Amateur Teleprinting Over Radio (AMTOR) are almost identical implementations of ARQ/Forward Error Correcting (FEC) technology that is intended to provide error-free, time diversity communications over HF radio. CCIR Recommendation 476-3, Direct-Printing Telegraph Equipment In the Maritime Mobile Service, specifies the SITOR handshaking protocol and codes.

The seven-bit Moore code is used in lieu of the five-bit Baudot or Murray code. In addition to the Baudot characters, the Moore code has three control characters. The only valid characters are those with bit combinations of three "1"s and four "0"s. The transmitting equipment first converts the teleprinting code to Moore code, and then converts the code to AFSK. In ARQ mode, after synchronization, the transmitter sends the converted characters in groups of three. The receiver converts the AFSK to Moore code, checks the incoming combinations, and rejects any that do not have the requisite three l's and four 0's. The receiver begins the sequence, again, by sending either an all-correct or send-again character back to the originating transmitter. The receiver converts the characters that it considers correct to a suitable form for driving an attached printer. In the FEC mode there is no handshaking. The transmitter sends (broadcasts) each Moore character twice to any station that is listening (redundancy). SITOR provides low error rate interference resistant operation. It is quite effective during conditions of fading and time diversity transmission/reception (assuming ARQ mode). However, this performance is provided at the expense of low throughput. As channel conditions deteriorate, ARQ mode throughput decreases because of required character repetition. FEC transmission becomes less reliable. At 100 Baud transmission rates, multipath interference is high enough to slur adjacent digital pulses. The resulting error would cause the ARQ system to make additional transmissions. In some installations fairly expensive equipment is required at each end of the transmission link. SITOR requires very stable (approximately 5 ppm) oscillators and transceivers at the sending and receiving stations. SITOR equipment is still used extensively on merchant vessels and thus provides to the Coast Guard a common resource over which to communicate with these vessels.

However, very little new development is occurring with respect to SITOR technology. Thus, although it provides a desirable "basic" technology, SITOR is being rapidly eclipsed by new HF technology. In particular, new generations of HF modems are supplanting ARQ/FEC technology as their costs decrease. Superior multi-tone time and diversity modems and associated receivers are becoming steadily cheaper.

8 COMMUNICATIONS CHANNEL MULTIPLEX/ACCESS

8.1 Overview

There are many established techniques for multiple users sharing a communications channel. These techniques combine the individual user signals into a composite baseband signal and are used in both satellite and terrestrial RF communications. Three major access methods are discussed in this section – TDMA, FDMA, and Code Division Multiple Access (CDMA). Within these access schemes, individual user accesses are separated through the one dimensional domains of frequency FDMA, time (TDMA), or the two dimensional frequency/time domain (CDMA). The complexity of the individual access schemes varies as access schemes may be statically pre-defined or dynamically defined (assigned) on demand [i.e., Demand Assigned Multiple Access (DAMA)].

8.2 Time Division Multiple Access (TDMA)

TDMA is an access technique based upon Time Division Multiplexing (TDM). TDM is a process whereby a common channel is time-shared by multiple independent signals. On input, TDM electronically switches the multiple input signals, so that periodically a sample of each signal occupies the common channel. On output, at a system's receiving end, decommutation circuits synchronize/sort the sampled signals. An overview of this scheme is illustrated in Figure 8-1.

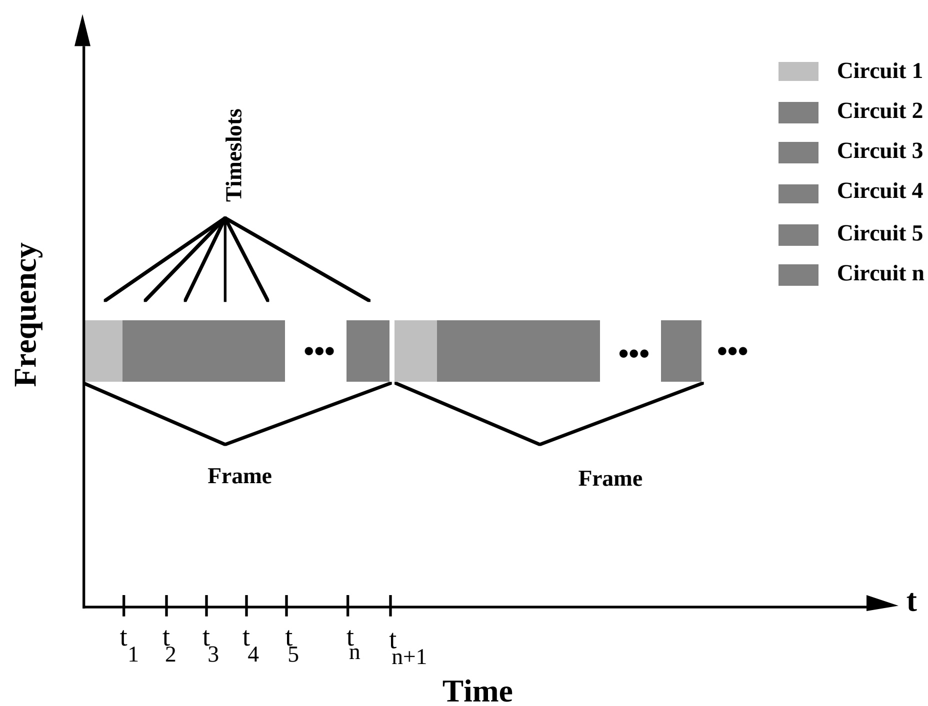


Figure 8-1. Time Division Multiple Access (TDMA)

TDMA is a technique for sharing a communications channel amongst a number of users. In this access scheme, a carrier frequency is divided up into sequence of time-slots. Each slot is a constituent component of a user circuit. The assignment (circuit), number, and size of the slots, as well as the composite channel bit-rate is implementation specific. Each transmitter is assigned a time slot, and the time slots are chosen to be non overlapping when they arrive at the receiver. Careful selection of the time slots is critical since the distances from the various transmitters to the receiver are not necessarily the same. Although signals may not be overlapping upon transmission differences in distance may cause overlapping upon reception. Guard times or dead bands are inserted between transmissions in order to alleviate the overlap problem.

The TDMA system configuration can be further broken down. Each transmitter is assigned a frame. However, each sender does not transmit at the beginning of its frame since overlapping at the receiver would occur. Instead, each transmitter (T1, T2, T3…) is assigned its time slot within the frame. The frames are grouped in combinations known as superframes. Each superframe has a collection of frames (perhaps, for instance, 64 frames). A typical hierarchy for a TDMA system is shown in Figure 8-2.

```
SUPERFRAME
    - Frame 1
            Time Slot 1
                    Preamble
                    User information
            Time Slot 2
                    Preamble
                    User information
                            •
                            •
                            •
            Time Slot j
                    Preamble
                    User information
    - Frame 2
            Time Slot 1
                    Preamble
                    User information
                            •
                            •
                            •
            Time Slot k
                    Preamble
                    User information
                    •
                    •
                    •
```

Figure 8-2. TDMA Hierarchy

As can be seen, a number of transmitting stations comprise each frame. Each transmitting station contains a preamble, and information (data) to be sent to other stations. The preamble includes a guard time before transmission is begun. Then a string of synchronization characters is transmitted that give the carrier sync recovery loops and bit timing recovery loops (in the ground station receivers) time to lock onto this RF burst. The end of the preamble contains a unique word that identifies both the transmitting and receiving stations. The high energy RF burst constitutes a PCM signal usually quadriphased modulated. The burst may be composed of several messages addressed to different receivers. In summary, TDMA possesses the following characteristics:

- TDMA allows several stations to use the same frequency band
- Transmission bursts from stations do not interfere (overlap)
- Transmissions from stations are separated in time
- Stations have access total channel capacity during separate time intervals and offers the following advantages:
  - Greatly reduces the effects of interference and intermodulation noise
  - Significant increase in channel capacity can be realized using TDMA techniques
  - Can achieve efficiencies in satellite power utilization of 90% or more compared to the 3 dB to 6 dB loss in power efficiency characteristic of FDMA operation
  - Similar bandwidth efficiency is obtained when accurate timing techniques are implemented
  - Great flexibility in planning channel sharing by multiple stations

It does, however, possess the following inherent disadvantages:

- Equipment may require large memory capability in which to buffer incoming data from other lines
- Requirement for sophisticated synchronization (e.g., strict burst timing to prevent burst collisions)
- Technical problems associated with time delay, system timing, frame rate selection, data storage, and error probability
- Economic (cost) factors associated with complex equipment

In spite of the current technical and economic issues associated with TDMA technology, these factors are diminishing over time as there exists and will continue to exist ongoing efforts in improving this technology, particularly with respect to satellite based implementations (e.g., onboard satellite processing, switched narrow-beam satellite antennas). Furthermore, there continues to be ongoing research in the areas of error probability, message delays, and unique-word detection associated with TDMA technology.

8.3 Frequency Division Multiple Access (FDMA)

This is simply the implementation of a narrow band channel when users have access to a collection of frequencies. FDMA is essentially the definition and implementation of frequency differentiated circuits within an allocated RF spectrum – an access technique based upon FDM. FDM allows a large number of channels over a specific range of frequencies to be transmitted as a composite signal. It is a process whereby a common channel is time-shared by multiple independent signals. In FDM, a range of transmission frequencies is separated into constituent narrow frequency bands. These bands represent individual channels. These separate frequency signals (channels) are multiplexed onto a single composite signal (carrier) for transmission over a common channel (the independent signals are separated in frequency as separate subcarriers) and are de-multiplexed at the receiver into its constituent subcarriers. An overview of this scheme is illustrated in Figure 8-3.

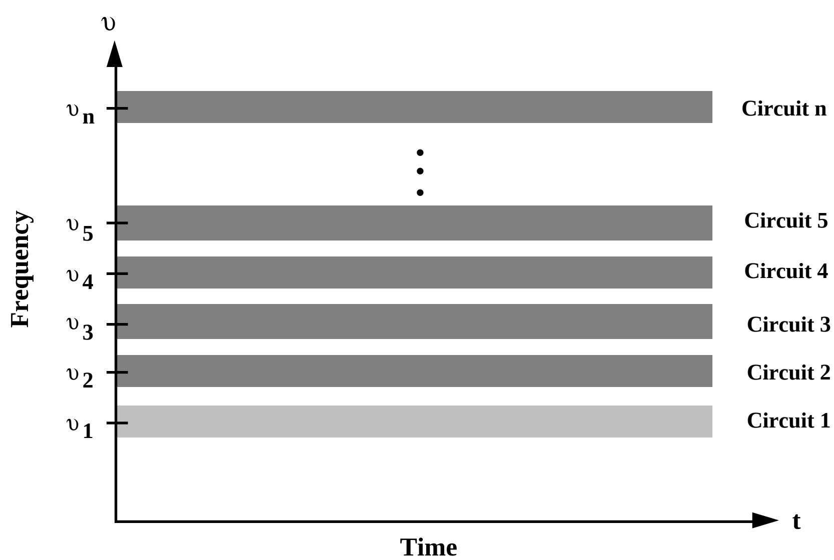


Figure 8-3. Frequency Division Multiple Access (FDMA)

In FDMA, each user is assigned a separate carrier frequency. Each user signal is allocated a separate, non overlapping, frequency channel usually accompanied by guard bands. This technique is called channel formatting. Intermodulation products are either accepted or minimized by appropriate frequency selection and/or reduction of input power levels. This multiple access scheme is used in satellite communications for multiple access of a single transponder by multiple uplink and multiple downlink stations. It is characterized by signals of constant envelope. Transmissions from individual users are extracted simply by filtering the composite carrier signal. Its main advantages are in its simplicity:

- No need for sophisticated synchronization processing (as required for CDMA and TDMA)
- Use of simple inexpensive equipment

Nevertheless, it possesses the following problems:

- Static determination of frequency band allocation, including guard bands – dynamic reconfiguration is difficult
- Multiplexing difficulties
- Intermodulation distortion, and adjacent channel interference - quality of transmission is highly dependent upon intermodulation distortion due to improper frequency selection and allocation

In spite of these difficulties, there will continue to be research in areas associated with FDMA technology – particularly in the areas of channel formatting, transmission filters, and single-channel dynamic demand assignment techniques. FDMA is currently being used with the INTELSAT IV and V series satellites. In addition, it is the principle access technique employed by current (analog) cellular communications systems

8.4 Code Division Multiple Access (CDMA)

CDMA is that form of multiple access used by spread-spectrum systems. This access scheme operates interactively with the spread-spectrum modulation techniques of frequency hopping and time hopping.

Spread spectrum modulation includes various modulation techniques which increase a signal's bandwidth beyond that required for carrying the information being sent. Band spreading is accomplished by one or more "codes" which are independent of the data. Synchronized reception "de-spreads" the signal and recovers the data. The fundamental principle of spread spectrum is that of distributing a relatively low dimensional data signal in a high dimensional environment. The modulator, using a fixed amount of power spreads it over all coordinates. This induces just a little interference in each coordinate. Alternately, the modulator places all the power into a small subspace, leaving the rest of the space interference free. The various wave form modulation techniques are:

- Frequency hopping. Information is modulated by a carrier, whose frequency is pseudo-randomly hopped over time. Narrow-band interference may collide with one frequency of the hop, but leaves the other frequencies untouched

- Time hopping. Data pulses or bursts are sent at pseudo-random times. The receiver listens only when the bursts are sent
- Direct sequence. This is also known as Pseudo Noise. A pseudo-random sequence of phase transitions is impressed on the data bit stream before or after the carrier is modulated
- Chirp. This is also known as Linear Frequency Modulation. Here a constant envelope contains a swept frequency

Spread Spectrum modulation possesses the following fundamental characteristics:

- Can be designed to minimize multipath effects
- Is relatively immune to interference and jamming
- Is capable of selective addressing (coding) for multiple users
- Is virtually undetectable by all but the most sophisticated monitoring
- Generally require sophisticated, expensive equipment
- Its underlying technology has primarily come from the DoD community

Spread Spectrum has found many applications in the area of covert communications. Furthermore, the Global Positioning System uses direct sequence Spread Spectrum techniques. It is currently being considered for applications such as mobile telephone and microwave communications in congested areas. It is a powerful tool that can revolutionize the whole philosophy of spectrum utilization. Although sophisticated Spread Spectrum equipment exists today, further technological refinements are still necessary in order to drive the relatively high costs associated with this equipment down. This economic consideration is of overriding concern which must be resolved before any serious consideration can be given to replacing conventional RF modulation systems with spread spectrum systems.

In summary, in a CDMA scheme each user accesses a wide band spread-spectrum channel according to its unique "code" which specifies in a pseudo-random manner the time-slot and frequency during which it transmits and/or receives transmissions. This concept is illustrated in Figure 8-4.

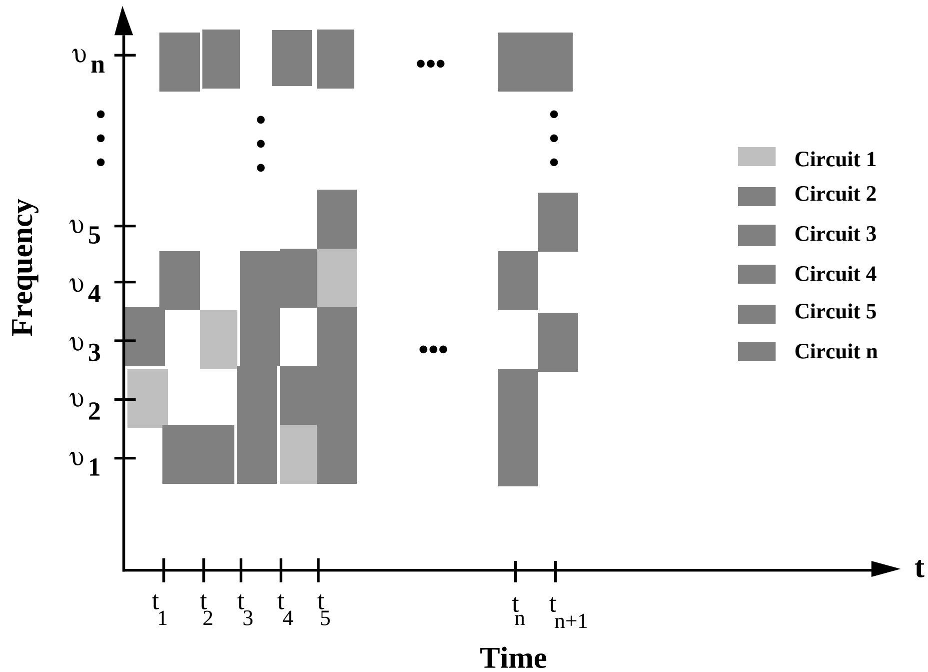


Figure 8-4. Code Division Multiple Access (CDMA)

9 COMMUNICATIONS NETWORKS

9.1 Switching Techniques

Two basic types of switching systems can be defined for computer communications networks transferring information: circuit and data. In circuit switching the complete physical path from the origin of a message information to its destination is allocated to the sender and receiver during the duration of the message transfer or until the "connection" is terminated by one or the other parties. This requires the need to set up the end-to-end connection before any information is transferred. During the transfer of the message data, any intervening network nodal entities (switches, in telecommunications networks) have no capability to buffer or manipulate the user message data. Once this "circuit" is established, data is transferred without any intervention on the part of the carrier network. For applications such as voice, which needs to transfer data via an effective continuous and constant "stream" rate (i.e., without occasional gaps in the continuous stream of data), this technique is generally the only manner in which voice data can be transferred if any manipulation (buffering, packetization) by the underlying network causes a non-continuous stream transfer from end-to-end.

The other main type of switching technique consists of two distinct variations – message switching and packet switching. The message storing system is also referred to as a "store-and-forward" system. In this type of switching, the entire message is transmitted as a unit to a directly "attached" node. Each intervening node in the network must receive a message in its entirety before forwarding it on to its destination. Although infrequently implemented, this technique is crucial in some applications. For example, when sending messages from one security domain to another. For instance, if a user in a "secure" subnetwork wanted to send a message to another user in another "secure" subnetwork which was connected by a non-secure public subnetwork, the overall network design could be based on the message switching architecture to meet their security requirements. That is, the "store-and-forward" node would be a gateway between the secure and non-secure subnetworks. After receiving a "secure" message from a user which needs to be passed to its destination via the non-secure subnetwork, the node would them encrypt the message before passing it over the non-secure network. Likewise, when the message reaches the other end of the non-secure subnetwork, the store-and-forward node in the other domain would decrypt the message and pass it on to the destination user.

A packet-switching system adaptively routes message segments rather than dedicating a channel to a transmitter and receiver before the communications takes place. The message is divided into packets. Each packet contains the desired data signal and address information. The system consists of a series of computer-controlled switching nodes connected by communications lines. As a packet enters a switching node that node determines the intended address and chooses a routing to the next appropriate node. At any point in time, many packets will be heading toward their intended destination.

Datagram and virtual-circuit are two technologies used in packet switching at the present. The more prevalent datagram technique allows for a packet to enter a system with routing decisions to be made at each node. The virtual-circuit technique calls for the path to be established between sender and receiver. The entire message uses this path. The same path may not be used for all packets in a message, but the path is established before a packet enters the network. The virtual-circuit is essentially a compromise between the datagram and a dedicated (inefficient) circuit.

Packet switching offers maximum utilization of available communications channels as a channel is occupied only during the transmission of a packet. There are no permanent circuits allocated exclusively for communications between specific communications sites. The virtual-circuit techniques have spawned a series of protocol standards has have been universally adopted. One major network protocol which defines procedures for setting up a call, transferring information, and ending the transmission is the so-called x.25 protocol as specified by the Comité Consultatif International de Télégraphique et Téléphonique (CCITT). Others include the widely used Internet Protocol (IP) and the OSI Connectionless Network Protocol (CLNP). Packet switching imposes some processing requirements on nodes within the network. First, they must have sufficient buffering capabilities, since incoming packets to a node must be queued prior to transfer to the next node. Also, a deadlock or lockup condition may occur when there are too many packets competing for available lines and the system exceeds its capacity. Another common problem occurs when a packet returns to a node for a second time if decisions at each node take it around a circle (looping). For complex networks, prior to the implementation of the network, extensive computer simulations may be necessary and heavily relied upon.

Continued research in many aspects of both packet switching and circuit switching will continue as new networking demands are constantly being imposed on existing physical communications assets. New protocols are being developed at a steady rate and to meet new requirements in the area of voice, full-motion video, facsimile, imagery need to be transferred over existing and newly introduced communications circuits.

9.2 Ethernet

"Ethernet" refers to the local area network standard developed by the Xerox Corporation. Standard Ethernet features a baseband transmission of 1–10 Mbps. It is fully specified by the IEEE 802.3 standard. The 802.3 standard is a superset of the original Xerox Ethernet specification as the original standard dealt exclusively with a 10

Mbps baseband rate over a 50Ω ohm coaxial cable. Today, it is also implemented at a 10 Mbps rate over unshielded twisted pair telephone type wiring (10Base-T). Other physical implementations include 1Base-5 (1 Mbps over twisted pair), and 10Broad-36 (10 Mbps over 75Ω ohm broadband coaxial cable). It is physically implemented in physical topologies ranging from a simple segment to branching networks of coaxial segments linked by repeaters as illustrated in Figure 9-1.

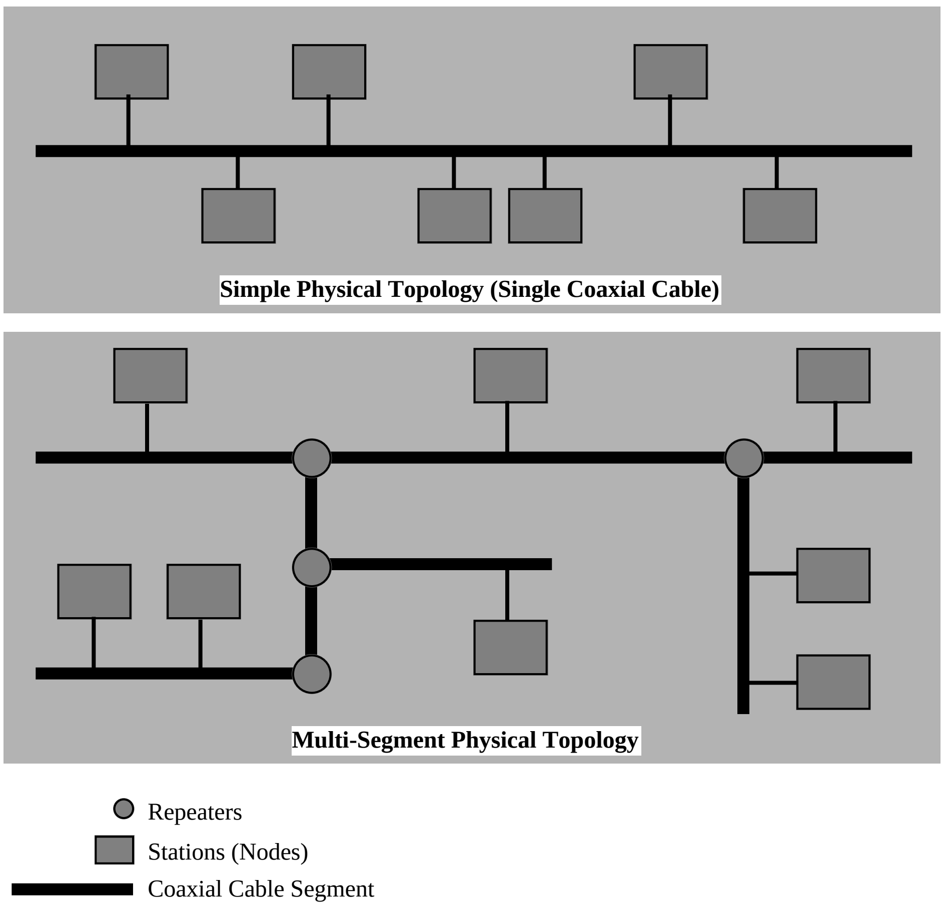


Figure 9-1. Ethernet Physical Topologies

From the viewpoint of the OSI layering abstraction, the Ethernet standard is implemented in two distinct layers: the physical and the lowest component of the data link referred to as the Medium Access Control (MAC) Sublayer. At the physical layer, Ethernet, transfers binary signals employing standard encoding techniques. At the MAC sublayer, Ethernet is implemented as a contention bus, performing a protocol referred to as "Carrier Sensing, Multiple Access With Collision Detection" (CSMA/CD). In this protocol, a sending station transmission is accessible to all stations on the network as all stations are continuously "listening". The unit of information sent is referred to as a frame. These frames contain the message data encoded as a variable length sequence of bits and both the identity of the sender and intended recipient of the data. Packet size ranges from 64 to 1518 bytes. Data is transmitted at a rate of 10 million bits per second. Figure 9-2 illustrates the layout of an Ethernet packet.

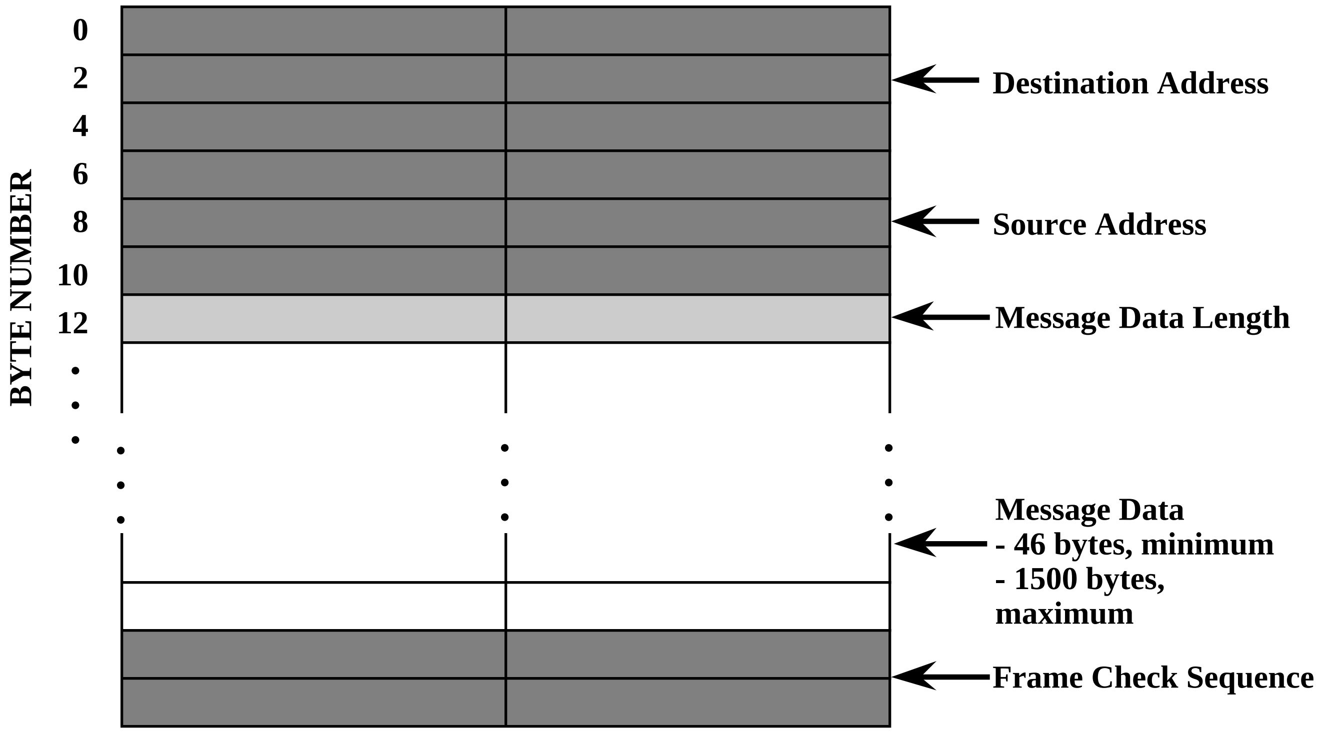


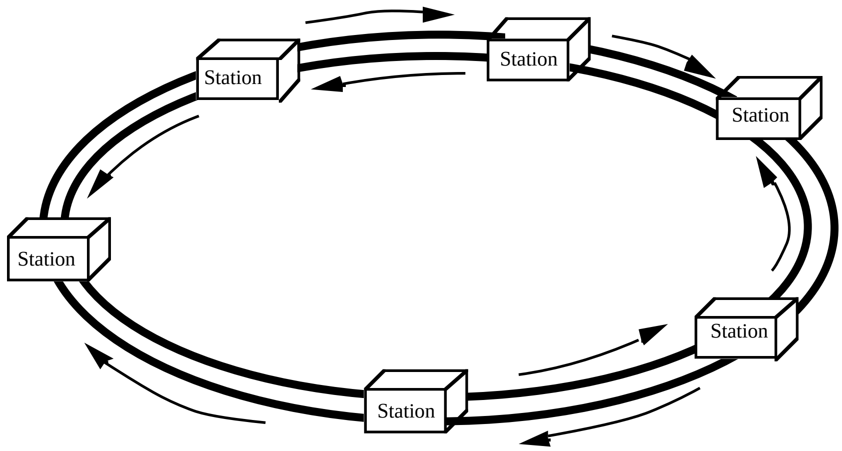
Figure 9-2. Ethernet Packet

The CSMA/CD protocol provides mechanisms to deal with packet collisions (from multiple stations sending a frame at the same time) and provides an error detection mechanism implemented through packet checksums. This protocol, however, does not perform any error recovery functions. It is left to an upper layer protocol to perform this function (e.g., IEEE 802.2 "Logical Link Control").

9.3 Fiber Distributed Data Interface (FDDI)

Fiber Distributed Data Interface (FDDI) is a collection of American National Standards Institute (ANSI) standards (ANSI X3T9.5) which specify a 100 Mbps fiber optic local area network. This is an order of magnitude faster than Ethernet. Aside from this speed advantage, FDDI can support more active users and more effectively uses the network bandwidth. For instance, an Ethernet network with just a handful of stations becomes heavily loaded when its bandwidth is 40% used. FDDI maintains excellent performance at even 80% bandwidth utilization. FDDI is based on token passing rather than the contention based design of Ethernet whose performance degrades rapidly under heavy load. In a token passing implementation, a token (a uniquely formatted bit pattern) continuously circulating in the ring. When circulating, no other data is transmitted in the ring. When a station is ready to transmit data, it must first "acquire" the token. Thus, this scheme effectively eliminates contention problems as only one station at any given instant is capable of transmitting data.

FDDI normally serves as a network backbone. It this role, FDDI is implemented as a dual, counter-rotating fiber optic ring, as illustrated in Figure 9-3.



Two Counter-Rotating Rings

Figure 9-3. FDDI Ring Configuration

In this configuration, stations are attached to two independent fiber rings. Data transmissions occur in opposite directions. One of these rings is denoted to be the "primary" ring. It is over this ring that data transmissions normally take place. The other "secondary" ring is normally idle. However, when a fault occurs in the network, the secondary ring is used automatically to work around this condition. When a fault occurs within a station, the "faulting" station is automatically detected and isolated by its immediate neighbor stations to its "right" and "left". These neighbor stations will internally connect their primary and secondary rings together. The secondary ring is used to automatically construct a complete ring. The two independent counter-rotating rings have now become a single ring. Likewise, should a break occur in the primary fiber optic cable, the adjacent stations will automatically detect the break and internally wrap the primary cable onto the secondary cable. These scenarios are illustrated in Figure 9-4.

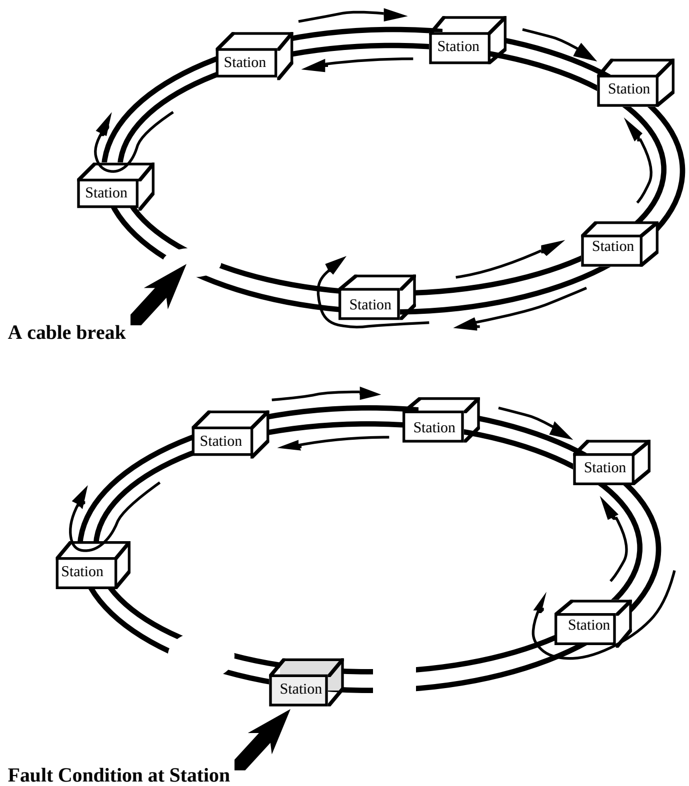


Figure 9-4. FDDI Fault Scenarios

The FDDI standards describe a layered network architecture which map into the OSI physical and the data link (bottom sublayer) layers. The FDDI standard is implemented by four distinct layers. These are listed below:

- Physical Media Dependent (PMD)
- Physical Layer Protocol (PHY)
- Medium Access Control (MAC)
- Station Management (SMT)

Figure 9-5 depicts the architecture of FDDI and its relationship to the OSI model.

OSI Layer	**FDDI Protocol Stack**	
Data Link (Layer 2)	**Logical Link Control (IEEE 802.1)**	
	Medium Access Control (MAC)	**Station Management (SMT)**
Physical (Layer 1)	**Physical Layer Protocol (PHY)**	
	Physical Media Dependent (PMD)	

Figure 9-5. FDDI Relationship to the OSI Stack

The FDDI physical layer consists of two components – the PMD and PHY components. PMD specifies the optical link parameters such as physical connectors, cables, optical frequency, and optical signaling levels. Normally, multi-mode 62.5 micron fiber optics are used to implement FDDI. However, if distances between stations are on the order of tens of thousands of meters, the more expensive (LASER versus diode light source) single mode fibers are used. PHY specifies the manner in which data is encoded/decoded and clocked. It also defines the 4-bit entities called "symbols' which are the elemental unit of transmission for FDDI. The PHY is the critical component of FDDI and is that particular feature of FDDI which makes FDDI unique in local area network technology. It is component responsible for performing the tricky signal synchronization between stations (due to the 100 Mbps rate). The FDDI "symbols" are used at this layer by adjacent stations to RAPIDLY exchange state information and the state of their connections.

The data link layer consists of the FDDI MAC component and IEEE logical link control protocols as defined by IEEE 802.2. MAC specifies the FDDI frame addressing/layout and token passing protocol.

The FDDI SMT component traverses both the physical and data link layers. Briefly, it performs the following functions:

- Manages the ring – detects and resolves ring faults
- Manages inter-station connections – ensures stations are connected according to FDDI "rules"
- Configuration management – maintains up-to-date configuration information for the other FDDI components
- Entity coordination – controls optical bypass relays (mechanism to bypass "powered-off" stations)
- Frame management – provides a method for "external" monitoring and control of the ring (e.g., station error statistics)

FDDI networks are particularly effective in relieving overload conditions of an Ethernet based network by being implemented to serve in the role of a backbone network tying together a collection of individual Ethernet based LANs. Current available technology permits RISC processor based bridges to route data between Ethernet LANs and FDDI backbones at rates up to 60,000 packets per second. FDDI is also a natural technology for the Coast Guard from a shipboard and security perspective. It is lightweight and resistant to electromagnetic interference. Furthermore, it is difficult for unauthorized users to tap into the network. FDDI is, however, a relatively expensive technology. For instance, although costs have been dramatically dropping recently, commercial FDDI interfaces cost on the order of $1000 today. This can be compared to an Ethernet interface which can be obtained for under $50. As far as future trends with FDDI are concerned, one significant effort underway is the FDDI Follow-On (FDDI-FO). This is ANSI's next local area network standard, designed to be an upgrade for FDDI users. It is scheduled for completion in the mid-90's and is being specified to operate between 600 Mbps and 1.25 Gbps.

9.4 Synchronous Optical Network (SONET)

The Synchronous Optical Network (SONET) was developed by the American National Standards Institute (ANSI). It specifies a physical digital interface over optical media. The SONET physical protocol consists of four sub-layers: path, line, section, and optical physical sub-layers, as illustrated in Figure 9-6.

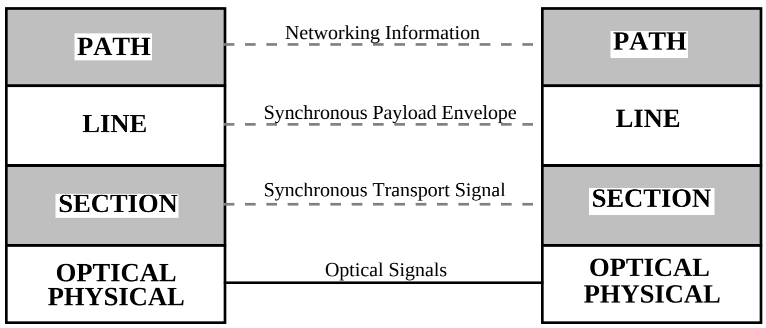


Figure 9-6. SONET Protocol Stack

The path layer is responsible for the exchange of network information between SONET multiplexing equipment. The line layer is responsible for transferring the so-called SONET "payload envelopes" (containing user data) between SONET nodes. It also provides synchronization and multiplexing functions to the path layer. The section layer functions to transfer the basic SONET signal component, the Synchronous Transport Signal (STS), over the optical media. It is responsible for actions such as signal framing and error control. Finally, the optical physical layer is responsible for optically transferring bit information across the physical media.

It defines a family of byte interleaved multiplexed signaling rates based on 51.84 Mbps. Currently, SONET supports carrier rates of 1, 3, 9, 12, 18, 24, 36, and 48 times the base 51.84 Mbps rate (respectively, 51.84, 155.52, 466.56, 622.08, 933.12, 1244.16, 1866.24, and 2488.32 Mbps). The base 51.84 rate is referred to as the Synchronous Transport Signal level 1 (STS-1). Each STS-1 frame is defined to be 125 ms in length. It is structured as 9 rows by 90 columns of 8-bit bytes. The bits of an STS-1 frame are transmitted row by row, from left to right with the most significant bit transmitted first. The initial 3 bytes of each row are overhead bytes for the line and section sub-layers of the SONET protocol. The remaining 783 bytes (87 columns-by-9 rows) constitute the STS-1 Synchronous Payload Envelope (SPE) of which the first column (9 bytes) is designated for "path overhead". This leaves 774 bytes available for payload (user data). Figure 9-7 illustrates the STS-1 frame.

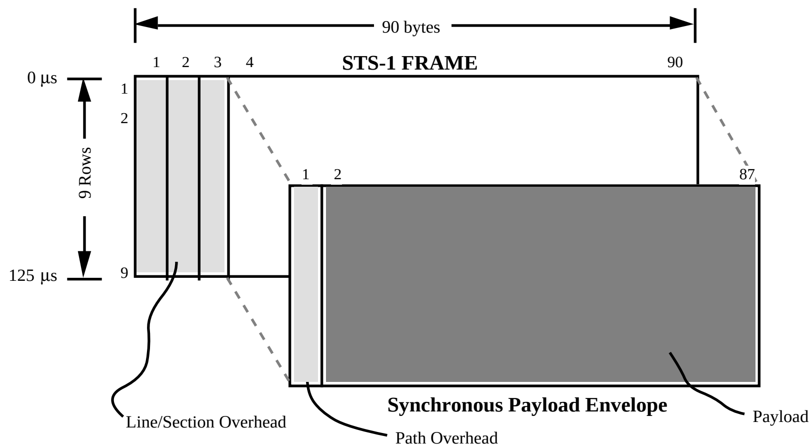


Figure 9-7. STS-1 Frame Layout

The SPE may be fully contained within a single STS-1 frame or may span multiple frames. Its beginning location within the envelope is specified by a payload pointer located in the "path overhead" portion of a STS-1 frame. A number N of 51.84 Mbps signals can be multiplexed to form a STS-N signal. The STS-N frame byte interleaves N STS-1 signals as illustrated in Figure 9-8.

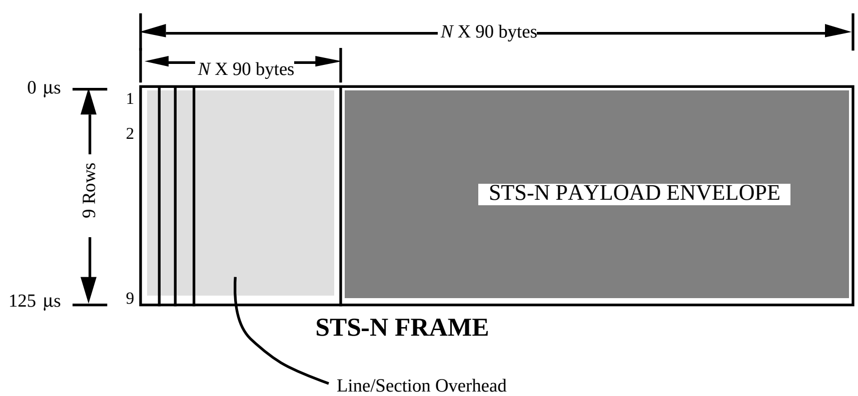


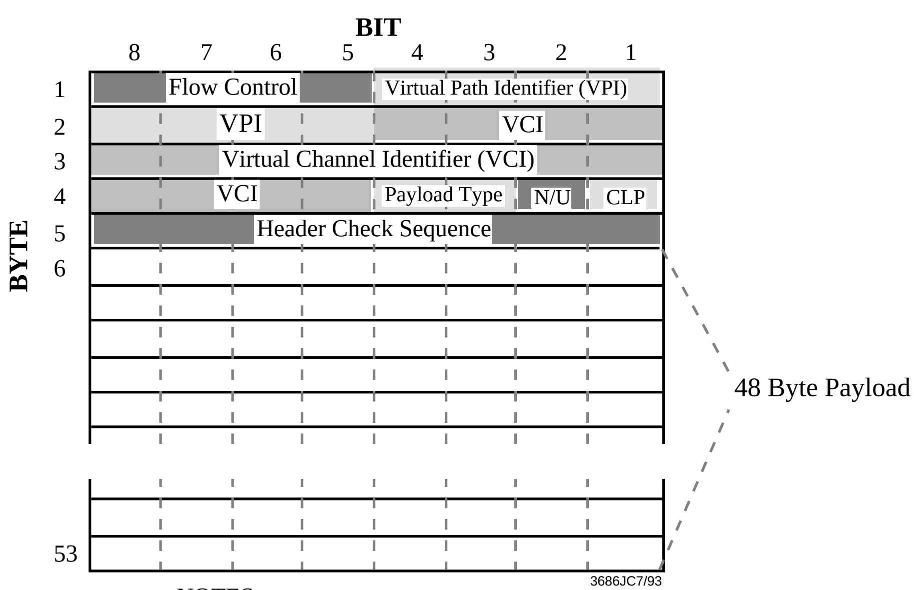
Figure 9-8. STS-N Frame Layout

Currently, SONET is a relatively expensive technology. Current implementations of SONET interfaces require integrated circuits employing technologies such as ECL, gallium-arsenide, and bipolar. These technologies are sometimes combined within a single multi-chip module.

9.5 Asynchronous Transfer Mode (ATM)

Asynchronous Transfer Mode (ATM) is a high performance multiplexing and switching technology. ATM standards are being developed by the ANSI and the CCITT. It has been adopted by the CCITT as the switching standard for Broadband ISDN (BISDN). ATM is characterized by its ability to transfer and route large volumes of all types of user data concurrently at very high speed over both local and wide area networks. The types of data which ATM can transfer include data, video, graphic images, and voice. Unlike multiplexed ISDN T1 channels, which separate these information types into specific fixed bandwidth channels, ATM freely mixes this information traffic and responds dynamically to changing bandwidth requirements. Thus, ATM provides the appropriate transport capability to each type of data. ATM combines the best features of current wide area network and local area network technology – packet and circuit switching. It has been designed to support **bursty** user traffic profiles.

ATM multiplexes and switches "cells" through networks on the order of gigabits per second. Each cell is essentially a small fixed length 53 byte packet of which 48 bytes are used to carry the data. User data is referred to as a "payload" in ATM terminology. Figure 9-9 illustrates the format of an ATM cell.



NOTES:
N/U - Reserved. Currently unused.
CLP - Cell Loss Priority; indicates whether cell has a lower priority and, therefore, can be "dropped" during overloaded conditions.

Figure 9-9. ATM Cell Format

The fixed length and size has been designed to promote rapid switching (designed for rates on the order of millions of cells per second). Switching information is contained within each cell. An ATM switch extracts routing information directly from the cell header. These switches are implemented by hardware which employs a self routing "switch fabric" whereby an incoming cell is routed to the appropriate output port without the intervention of any software processing. This hardware based switching allows this high order switching throughput. A simple example of a self switching "fabric" is illustrated in Figure 9-10. In this example destination addresses are encoded in a three bit field (i.e., 8 possible destinations) and the switching "fabric" is implemented with binary nodes (2 inputs, 2 outputs).

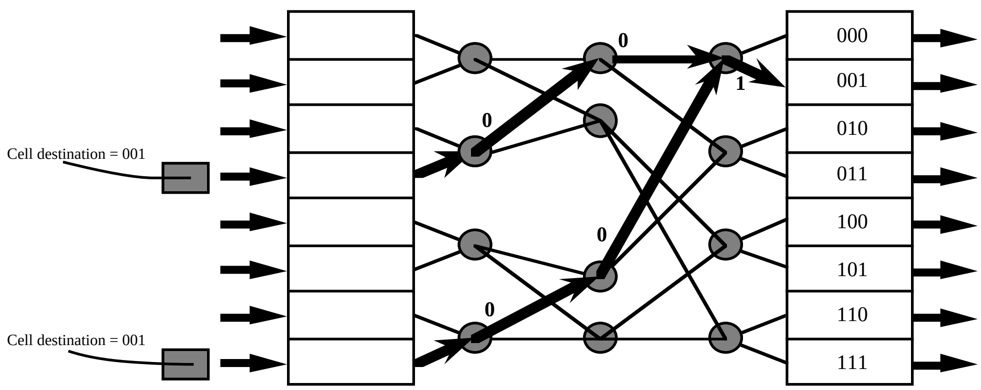


Figure 9-10. Self-Switching Fabric Example

When a cell reaches a node, the node will route the cell "up" if the first bit of the address is 0. Otherwise, it is routed "down". Also, the address field is shifted to prepare it for the next node (Note that the address field will have been shifted back to its original configuration by the time it is output from the ATM switch).

As far as the physical transport systems is concerned many different approaches have been discussed. One early scheme specified continuous "stream" transmissions of the 53 byte ATM cells. However, the physical transport scheme selected for ATM is SONET in the United States and the European CCITT Synchronous Digital Hierarchy (SDH). Both are optical standards defining the manner in which data can be transmitted at rates up to 2.488 Gbps. Figure 9-11 illustrates, at a very high level, how user data is transferred from user to user.

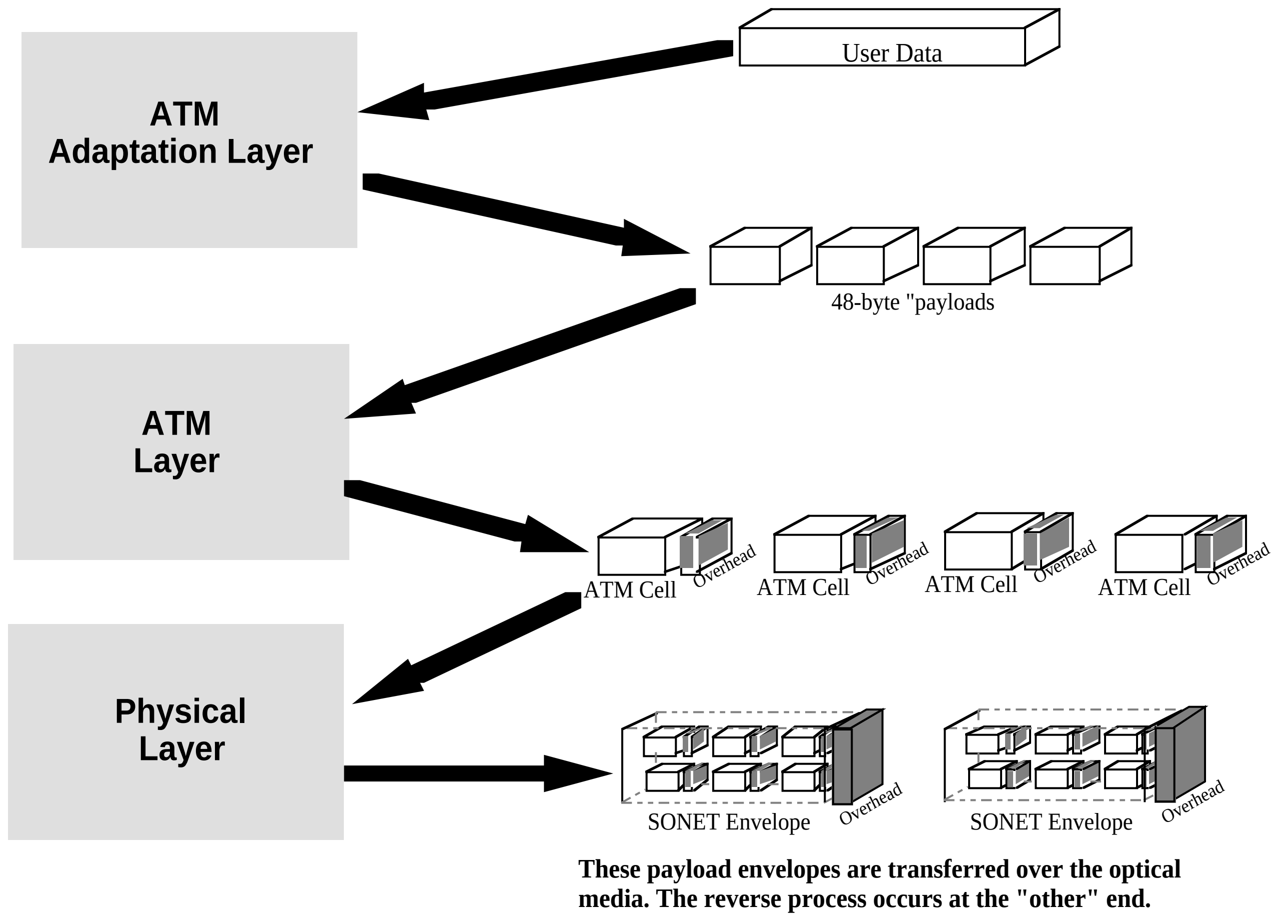


Figure 9-11. End-to-End ATM Data Transfer Service Overview

Two end-to-end users communicating over an ATM network first establish a bi-directional virtual circuit between them in a manner similar to standard packet switched networks. The bandwidth of these virtual circuits, ATM Virtual Channels (VC), are negotiated at set-up time as required for the end-to-end communications needs (e.g., video, voice, data, imagery). Each VC describes the unidirectional transfer of ATM cells. All cells which are part of this channel are identified by a common Virtual Channel Identifier (VCI). ATM also provides a provision to bundle a set of VCs. VCs can be associated with a Virtual Path Identifier (VPI) value. This may be used in applications such as teleconferencing where the information associated with full-motion video is assigned to VC "X" and the corresponding voice data is assigned to VC "Y". Figure 9-12 illustrates this concept

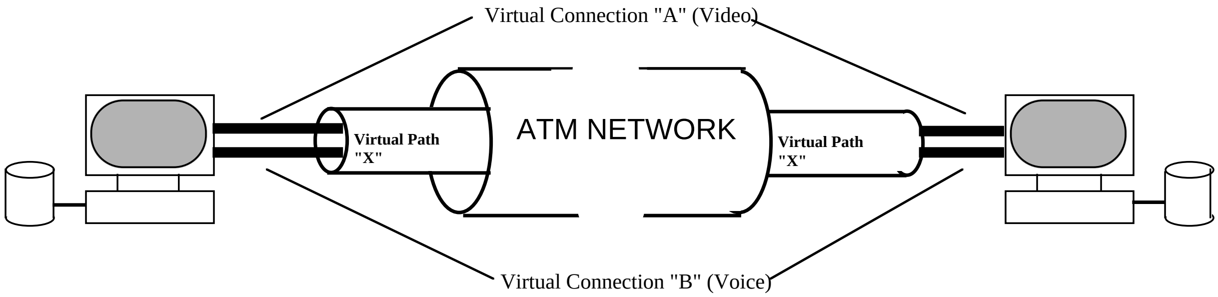


Figure 9-12. ATM Virtual Path/Virtual Channels

ATM essentially maps into the Open System Interconnect (OSI) Data Link protocol layer. The core cell relay protocols map closely with the functionality of the MAC and the Logical Link Control (LLC) components of standard local area networks. The ATM protocol stack architecture is similar to that of the ISDN in that there exists two separate stacks – one for the "user plane" and one for the "control plane" (signaling). This architecture is illustrated in Figure 9-13.

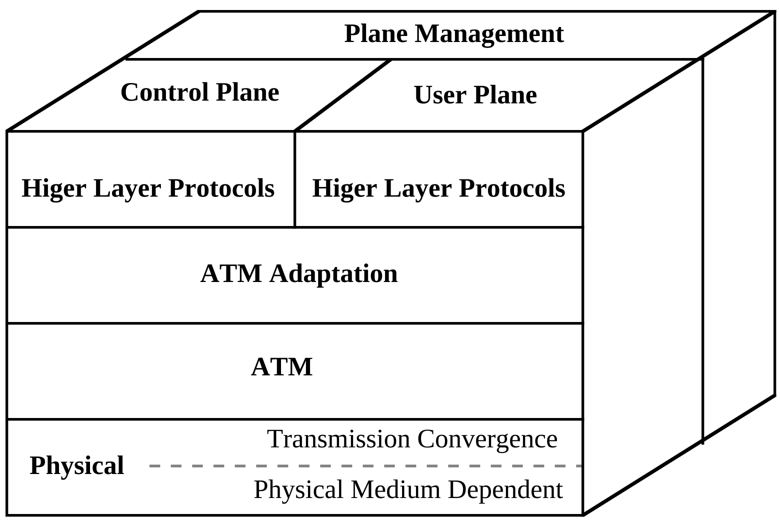


Figure 9-13. ATM Protocol Architecture

The ATM Adaptation Layer provides service dependent functions to the adaptation layer service user (i.e., the layer above it). Here user information is mapped into ATM cells (segmentation and reassembly). Transmission Control Protocol/Internet Protocol (TCP/IP) would be the equivalent service layer(s) in conventional local area networks. The Physical Layer consists of two distinct components – the Physical Medium (PM) dependent sublayer and the Transmission Convergence (TC) sublayer. The PM sublayer provides those media dependent functions such as bit alignment and opto-electrical signal conversion. The TC sublayer performs those functions necessary to transfer the ATM cells over the physical media. As mentioned previously, this functionality is provided in the U.S. by the SONET. The ATM layer performs cell multiplexing/demultiplexing. On transmission, cells from separate virtual paths and channels are combined into a composite flow of cells. Likewise, on reception, cells are extracted from the composite flow of cells and directed to the appropriate virtual channel or path. At ATM switches, VPIs and VCIs are translated into a new VPI and/or VCI value (ATM dynamically establishes and tears-down internal connections as required dynamically).

In the very near term, aside from its specified use for BISDN, ATM will find its way into congested Local Area Network (LAN) environments. The last five years have seen the explosive growth of LANs. In conjunction with the rapid growth in the raw numbers of LAN stations, we have seen the incredible increase in the computing powers of these LAN attached stations. This has created an exponential increase in requirements for LAN bandwidth. Recently, as LANs have become congested, backbones were created to alleviate the congestion. As the number of LANs attached to the backbone increases, the backbones themselves have experienced congestion and additional backbones are installed. However, there reaches a saturation point (due to the incremental increases in packet delay times across bridges), when the addition of another backbone brings forth no benefit. This is illustrated in Figure 9-14.

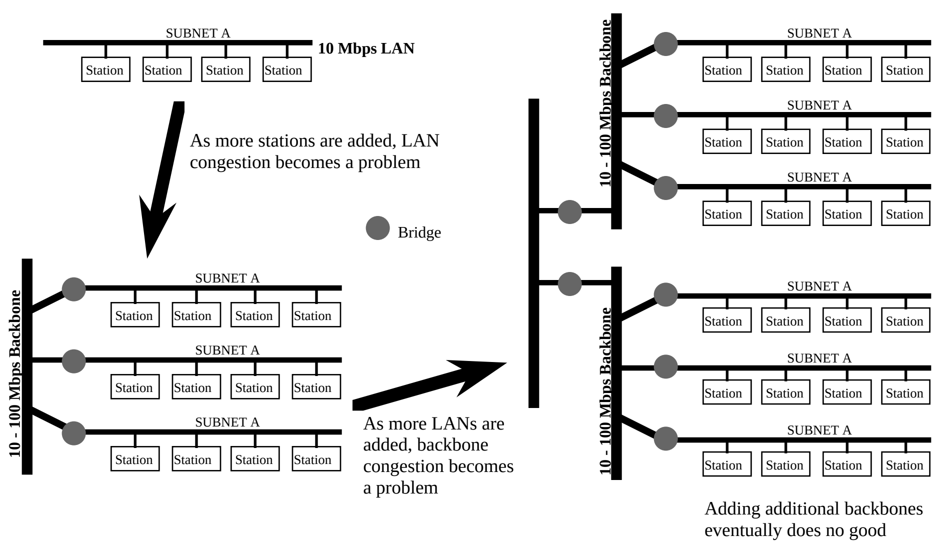


Figure 9-14. Evolving LAN Congestion

ATM will solve this LAN congestion problem by eliminating the need for backbones by consolidating all inter-LAN network traffic through an ATM switch. Furthermore, ATM will solve not only the congestion problem but also the many logistical problems associated with rapidly growing networks. That is, it will alleviate the problems experienced by network administrators managing the continuous physical movement and re-configuration of network assets. ATM separates networks physically and logically. Physical locations are completely independent to a station's network address (i.e., IP address). LANs are connected through an ATM switch via a virtual connection. A LAN which has been partitioned physically for whatever reasons need not have their network addresses re-configured. Stations on separate LANs have no knowledge of the ATM switch physically separating them. Figure 9-15 illustrates how ATM solves both the network congestion and physical re-configuration problems being experienced today by collapsing bandwidth limited interconnecting backbones into a single high-speed switch.

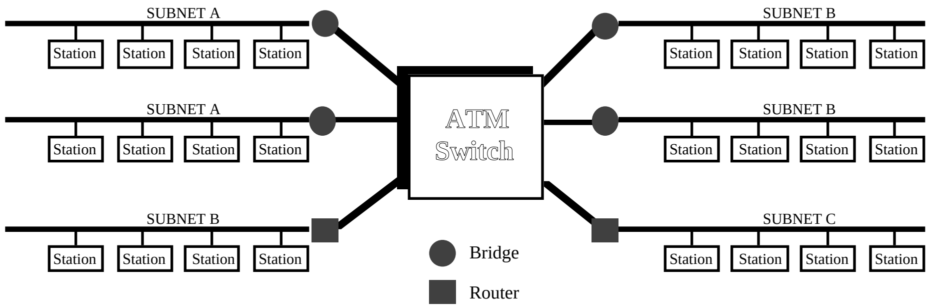


Figure 9-15. ATM-Based LAN

10 INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

10 INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

10.1 Overview

ISDN is a digital end-to-end network which supports a wide range of services, including voice and non-voice services. These services include digital voice, video, facsimile, and computer-to-computer file transfers. ISDN is defined in a series of specifications by the CCITT. Access by the ISDN user is limited to a set of standard multipurpose customer interfaces.

With ISDN digital terminals, including telephones, create signals. These signals include both the message itself and control and supervision information, and they are transmitted end-to-end in digital format. ISDN allows for signaling protocol which greatly enhances the level and type of services that vendors will be able to offer to telecommunications users. Because the ISDN message and its control signaling travel via discrete paths, vendor services can be accessed both at the initiation and during the message session. Additionally, given the broad range of signaling capability, bandwidth can be allocated as needed to a varying demand load on a dynamic basis. This allows for a more efficient use of facilities by the users as well as the vendors.

Two established services have been defined by the CCITT. The first (Basic Service) is a 144 Kbps channel that is subdivided into two 64 Kbps message channels and a 16 Kbps signaling channel. Each message channel can carry either a voice signal or a data signal. Each channel can be switched and controlled independently, based on signaling carried in the 16 Kbps channel. In addition, the 64 Kbps message channel may further be sub divided using a premises-provided mechanism, but the control signaling applies only at the 64 Kbps level. The second (Primary Service) is either a 1.536 Mbps or 1.984 Mbps channel subdivided, respectively, into:

- 23–64Kbps message channels; 1–16 Kbps signaling channel (1.536 Mbps)
- 30–64Kbps message channels; 1–16 Kbps/sec signaling channel (1.984 Mbps)

The 64 Kbps channel has been labeled the B ("bearer") channel and the 16 Kbps channels have been labeled the D ("delta") channel. Table 10-1 lists the various channel types which have been standardized for ISDN.

Table 10-1. ISDN Channel Types

TYPE	RATE	APPLICATION
A	4 kHz analog	telephone
B	64 Kbps digital PCM	voice and data
C	8 or 16 Kbps digital	-
D	16 or 64 Kbps	out-of-band signaling
E	64 Kbps digital	internal ISDN signaling
H	384, 1536, or 1920 Kbps digital	-

Thus, the Basic Service groups together 2 B channels + 1 D channel. Likewise, the Primary Service groups together 23 or 30 B channels + 1 D channel. This is illustrated in Figure 10-1.

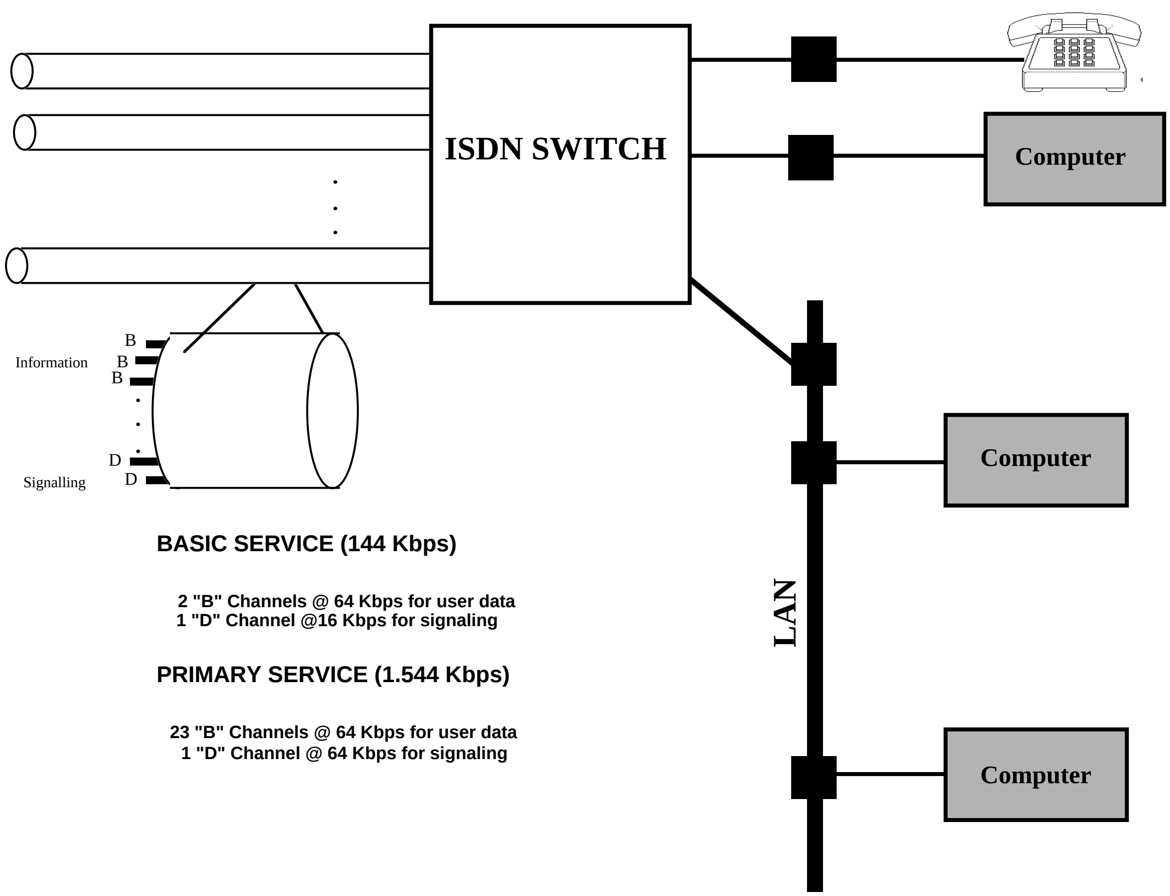


Figure 10-1. ISDN Architecture

The CCITT reference model for ISDN uses the following terminology:

- **TE1** (Terminal equipment, type 1). An ISDN compatible terminal device (e.g., computer, facsimile).
- **TE2** (Terminal equipment, type 2). A non-ISDN compatible terminal device (e.g., analog telephone).
- **TA** (Terminal adapter). A interface adapter for connecting one or more non-ISDN compatible terminal device, to the network performing any necessary protocol conversions.
- **NT1** (Terminal termination 1). A device on the telecommunications carrier's side of a connection performing any necessary signal conversions.
- **NT2** (Terminal termination 1). A device on the user's side of a connection performing switching and multiplexing a number of ISDN compatible terminal devices.
- **LT** (Loop/line termination). The termination of a line at the telecommunications carrier's central office.
- **ET** (Exchange termination). The interface to the telecommunications carrier's local exchange switch.
- **R**. A logical reference point between a non-ISDN terminal and a terminal adapter.
- **S**. A logical reference point between an ISDN terminal or terminal adapter and an ISDN network termination 2.
- **T**. A logical reference point between the user and the telecommunications carrier (between the network termination 1 and 2).
- **U**. A logical reference point between a the user's network termination and the user's line/loop termination.
- **V**. A logical reference point between the exchange termination and the line/loop termination.

Figure 10-2 illustrates the CCITT's ISDN reference model.

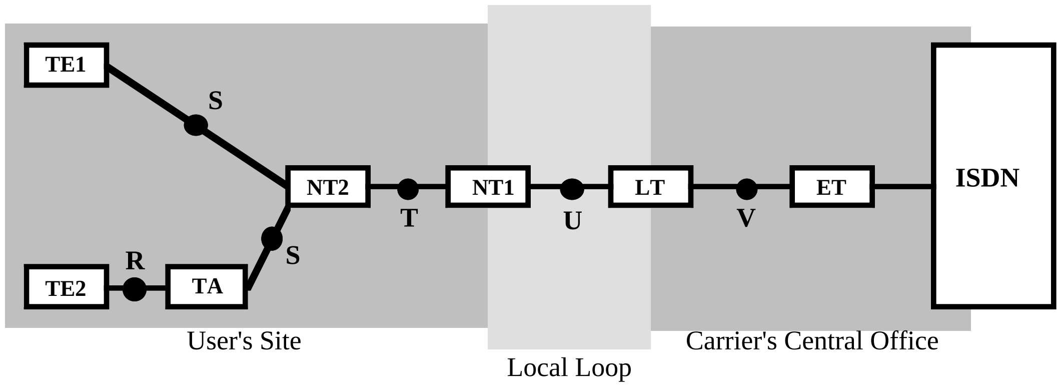


Figure 10-2. CCITT's ISDN Reference Model

ISDN technology provides tremendous advantages. This can be summarized as follows:

- ISDN offers major cost reductions to the carrier. Commercially speaking, AT&T communications, and eventually the consumer, will stand to gain due to the following benefits:
  - The reduced need for analog and digital transformation
  - The reduced call setup times because of discrete channel signaling
  - The reduced call holding times for data transmission at 64 Kbps
  - Increase network use by up to 10X. Local telephones companies will be able to Increase significantly the use of their existing cable plant. (ISDN will reduce the need for common telephone company central office equipment and maintenance needs)
- ISDN will cause a change from private-line services to switched services for both voice and data applications
- Because of the highly flexible but standard interfaces of the ISDN, future ISDN terminals will have a longer useful life, will be highly portable with universal access nationally and internationally, and will have access to a wide set of vendor-provided services to enhance terminal functionality
- For data terminals and computer front-end interfaces, the simplified and faster call set process of the ISDN, coupled with its enhanced signaling capabilities, will introduce a new family of data applications

- ISDN will enhance the user's ablate to implement new applications on a cost-effective basis, including various video and graphics capabilities
- ISDN will provide the ability to allocate network bandwidth dynamically (More efficient use of communications capabilities)

For the Coast Guard, this means applications such as integrated voice/data Electronic mail and video conferencing. Furthermore, it will permit communications networks to be consolidated by combining hardware and software from many vendors into one central network. In spite of these inherent advantages, ISDN is still a largely unfulfilled promise. Much research, lobbying, and salesmanship remains before ISDN evolves. Current problems are:

- Conversion Cost. The immediate expense and logistics of making terminals, phones, Private Branch Exchanges (PBXs) and other telecommunications gear compatible with the ISDN is a problem. For commercial carriers such as AT&T and MCI, this means the costly and time consuming upgrades of their analog and electromechanical switches.
- Equipment Cost. Although costs have dramatically dropped over the past few years, ISDN equipment is still not cost effective with current analog phones, modems, and LAN adapters. An ISDN interface for computers range from $1000-1500. On the other hand V.32bis/V.42bis modems bundled with error correction/data compression can be purchased for $400 and a LAN interface to a computer can be obtained for under $200. Furthermore, as far as phones are concerned, ISDN phones are only now approaching the costs of phones for digital PBXs.
- Full Accessibility. The benefits of ISDN are not observed, as ISDN is still not offered across a majority of the geography pertinent to the user communities. Typically, only large metropolitan areas have ISDN capabilities. As the number of potential users of ISDN equipment is denied access to this technology, the demand for equipment will be kept down which will tend to keep prices high.
- Integration. The simultaneous operation of an ISDN and non-ISDN environment has still not been adequately addressed.
- Interoperability. Although the major ISDN manufacturers (AT&T, Northern Telecom, Siemens, etc.) have recently agreed to a standard called National ISDN 1 to solve the interoperability problem plaguing ISDN in this country, some current end-user products will not be upgraded to this standard.
- Cost/Performance Competition. PBX manufacturers have been slashing prices and adding more features to their products.

- Data Throughput. Until developments with broadband ISDN makes its way to the marketplace, ISDN is too slow even with today's compression technology to handle some hot new communications applications such as full-motion video.

Undoubtedly, ISDN will spawn further technological advances. One can easily anticipate the development of centrally located digital switches capable of servicing thousands of users. Furthermore, ISDN will likely help push the development of very high speed communications circuits based on satellite, microwave, and fiber optic technology.

10.2 Basic Narrow Band ISDN and the OSI 7 Layer Model

The abstraction in common use today for describing or specifying communications between systems is the so-called Open Systems Interconnect (OSI) 7 Layer Model. This is fully described in Section 13, *Computer Systems Architecture Concepts* . ISDN deviates somewhat from the model as illustrated in Figure 10-3.

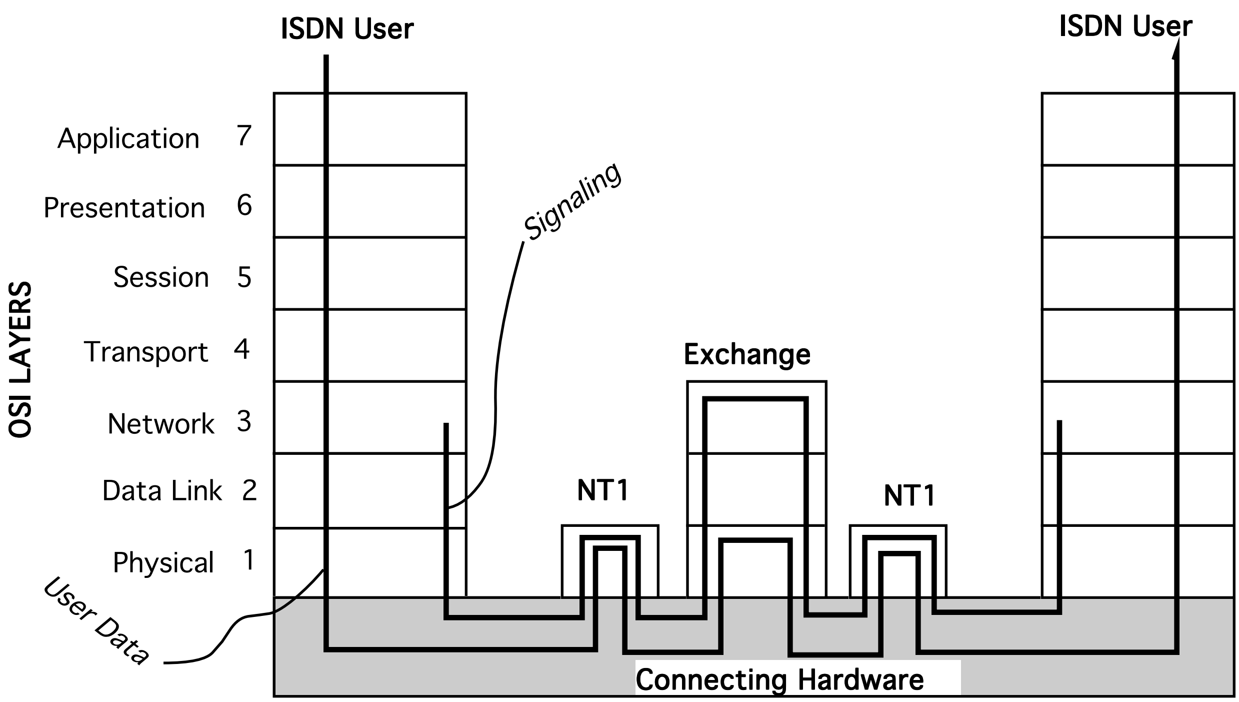


Figure 10-3. ISDN and the OSI Stack

Of significance is the fact that the user data (voice data, imagery, etc.) is only "transformed" at intermediate ISDN nodes such as at terminal termination points (NT1) even though control information has three defined functional layers. They key feature of ISDN is the logical separation of signaling information and user data. It can be looked upon as consisting of two subnets – a switched informational subnet and a signaling network.

10.3 Broad Band ISDN

The Basic Narrow band ISDN discussed in the previous sections deal exclusively with ISDN based on 64 Kbps "B" channels and 16/64 Kbps "D" channels. For many applications, substantially higher bit rates are becoming necessary. The provisions for data channels exceeding 64 Kbps is referred to as BISDN. Provisions for what actually constitutes BISDN vary substantially. Nevertheless, it suffices to state that BISDN will use ATM and will provide the following channels:

- H11 channel @ 1.536 Mbps
- H12 channel @ 1.920 Mbps
- H21 channel @ 32.768 Mbps
- H4 channel @ 132 – 138.24 Mbps (integer multiple of 64 Kbps)

11 CELLULAR MOBILE TRANSMISSION

11.1 Overview

Today's Cellular Mobile Transmission systems (henceforth referred to as "cellular radio") have over the past decade evolved from its implementation infancy into the dominant means by which mobile communications are effected today. It is not a new advanced technology. Rather, it is an "organization" of existing technology into a communications architecture providing communications services on a much grander scale. Cellular radio was implemented without any fundamental technological leap forward. It was implemented simply by reapplying existing communications resources. The core technology involved is still analog frequency modulation. Its innovation was based on the concept of "cells".

The cellular architecture represented a radically different approach to structuring a radio network. It is a system-level concept independent of radio technology. As shown in Figure 11-1, the cellular concept is exceedingly simple. Rather than utilizing a single centralized high powered transmitter broadcasting over a very large area, it rather specifies wide area coverage through the use of many low powered transmitters (base station) each of which is specifically designed to service a small geographic "cell". Thus, a large metropolitan area is segmented into a large number of small "cells."

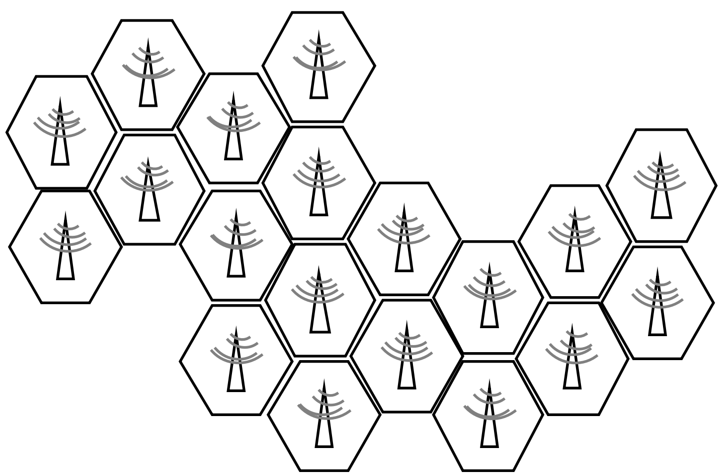


Figure 11-1. Basic Cellular Architecture

Due to interference between mobile radios operating on the same channel in adjacent cells, adjacent cells no not use every available frequency in every cell, as shown in Figure 11-2. To avoid interference, several cells are skipped before the same frequencies are reused.

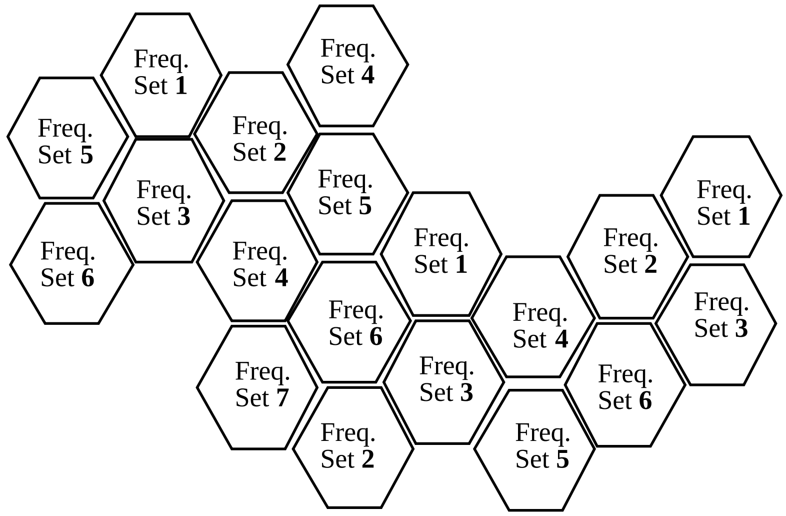


Figure 11-2. Frequency Separation

Frequency interference is not related strictly to the distance between cells but rather to:

(ratio of distance between cells)/(radius of the cells)

For instance, if a cellular network of 32 km radius cells permitted reuse of frequencies without interference between cells separated by 64 miles, then a cellular network of 16 km radius cells will allow reuse of frequencies at inter-cell distances of 32 km. This implies that for each cellular area reduction of 50%, the number of circuits will increase four-fold (assuming cell area equals Pr^2).

The final key architectural component of the "cellular" architecture is the concept of the "hand-off". When a mobile radio user transitions into another cell, the cellular system automatically "hands off" control of the mobile radio to the new base station. By providing a continuously monitored control and switching capability, the cellular system automatically "switches" communications from the old cell to the new cell automatically without affecting the communications. In summary, the key components of the cellular architecture are as follows:

- Frequency reuse
- Low power transmitters and small coverage zones or cells
- Hand-off and central control

Operationally, mobile units communicate by FM radio to the base stations, which in turn are hooked up via land lines to a mobile-telephone switching office. The office routes calls to other base stations via the main telephone lines. Standard cellular communications are narrow band systems. In this conventional architecture, cells are assigned their own frequency bands so that interference from adjacent cells is avoided. That is, the total spectrum allocation is separated into a collection of narrow radio channels defined by radio frequency. The bands can be assigned in a number of patterns. This is currently implemented through an FDMA scheme. For mobile units, this implies that it must be capable of tuning into all available frequencies and must be under complete control of a base station at all times. Furthermore, as transmission occurs continuously in both directions, both the mobile and base station radios must be capable of full-duplex operation.

11.2 Current Operational Characteristics

Today's cellular technology theoretically can handle large numbers of users that can be handled in the entire service area by reducing cell size with a corresponding reduction in the interference range around each cell (as discussed above). No longer is cellular radio an infant technology. Both user equipment and subscription rates have dropped significantly in price and there are now an estimated 10 million cellular phone users nationwide. The quality of service has improved significantly as the frequently experienced garbled and dropped transmissions, which were typical when cellular telephone networks went on-line in 1983, have now been substantially overcome through improved software and equipment.

11.3 Immediate Trends

There are currently multiple second generation (digital) cellular radio standards off the drawing boards ready for implementation. Access schemes proposed for these second generation systems include both – TDMA and CDMA mechanisms. Especially for the non-continuous burst mode transmission characteristics of TDMA based systems, the mobile radio can be significantly simplified through the elimination of duplexer circuitry. Duplexer circuitry can be replaced by a fast switch to toggle between the transmitter and receiver. This also implies that a single chip transceiver can be implemented for these digital cellular systems without the concern for cross talk which might be faced with analog systems. The ultimate result of this will undoubtedly be the reduction in both size and cost of the mobile communications unit. Furthermore, there has been a strong effort by the major cellular carriers to initiate a packet data network based on modified cellular networks before the end of 1993. Thus, as cellular technology is already evolving to its second generation (digital) characterized by low-cost and lightweight digital packet oriented networks, Cellular Mobile Communications appears to be a strong candidate for implementing at low-cost marine communications in harbor/inland waterways.

11.4 Future Trends

Cellular radio is now evolving into its second generation. Though more mature technically than present counterparts, it still exhibits significant shortcomings. It is still primarily geared towards telephone communications. The current control architecture and the RF bandwidth efficiency is still insufficient to meet the integrated networking characteristic of the future and the demand by the mass-market to access large volumes of data available via the integrated supernetworks of the future.

12 SATELLITE COMMUNICATIONS

12.1 Overview

Satellite communications has successfully competed with undersea cable for long distance transoceanic communications service. Today, the bulk of transoceanic communications is maintained by satellites. It has brought communications capability to rural and remote areas previously relying exclusively on HF radio links. Data, telephone, and television signals comprise the majority of the information relayed. Most satellites used today are placed in synchronous orbit fixed in space 36,000 km above the equator, maintaining a constant geographic position with respect to the earth (geostationary). Coverage of the earth's polar regions is provided by polar-orbiting satellites in which tracking (moving) antenna are necessary.

12.2 Frequency Use

Communication satellites use frequency bands (using military frequency designations) in pairs, one band for the uplink and the other for the downlink. All satellite frequency pairs assign the higher frequency to the uplink. The reason for this stems from the fact that the higher frequency suffers a greater degree of spreading loss compared to its lower frequency counterpart. The historical assumption set into place by satellite implementors has been that earth stations have greater power and antenna capabilities than a satellite (most are still powered by solar-cells). Presently, many systems have paired the 4 and 6 GHz bands (G-band) due to earlier allocation and the fact that this frequency band exhibits excellent propagation characteristics. In the 4 to 6 GHz band, synchronous satellites are assigned an orbital spacing of 4 degrees, but a 2 degree spacing is gradually being implemented.

Most recent satellites must operate at higher frequency bands since the G band is filling up rapidly. J band satellites use a 12 GHz downlink and a 14 GHz uplink with a 3 degree spacing. Direct-to-the-home broadcasting satellites television (TV) require a 17 GHz uplink and a 12 GHz down link. Other bands implemented include D band, J band, a special Direct Broadcasting Band (DBB) and a few more over 10 GHz. Table 12-1 lists some significant letter designations for satellite frequency bands.

Table 12-1. Satellite Frequencies

BAND	FREQUENCY RANGE	DIRECTION	TYPE
4 GHz	3.7 – 4.20 GHz	Down Link	Commercial
6 GHz	5.925 – 6.425 GHz	Up Link	Commercial
12 GHz	11.70 – 12.20 GHz	Down Link	Commercial
14 GHz	14.00 – 14.50 GHz	Up Link	Commercial
19 GHz	17.70 – 21.20 GHz	Down Link	Commercial
29 GHz	27.50 – 31.00 GHz	Up Link	Commercial

Table 12-1. Satellite Frequencies continued

BAND	FREQUENCY RANGE	DIRECTION	TYPE
7 GHz	7.250 – 7.750 GHz	Down Link	Military
8 GHz	7.900 – 84.000 GHz	Up Link	Military
20 GHz	20.200 – 21.200 GHz	Down Link	Military
30 GHz	30.000 – 31.000 GHz	Up Link	Military
20 GHz	20.200 – 21.200 GHz	Down Link	Military
44 GHz	43.500 – 45.500 GHz	Up Link	Military

Each synchronously orbiting satellite is assigned a position and a frequency band. With the 4/6 GHz band, a 500 MHz wide spectral assignment is made employing 12 transponders (receiver-to-transmitter). Each transponder uses 36 MHz with a 4 MHz guard band of the total 500 MHz. An additional 12 transponders can be employed when the same frequency band is "reused" and orthogonal polarization (vertical and horizontal polarization) takes place.

Multiple access of a single transponder by multiple uplink and multiple downlink stations is necessary to expand communication satellite system capabilities. Various multiple access techniques exist, such as FDMA, TDMA, and CDMA.

12.3 General Satellite System Characteristics

Numerous satellite systems employ the above mentioned frequency bands and expanding system technologies of frequency reuse, modulation schemes (QPSK, PCM, DM), and multiple access. A few systems deployed at present include, MARISAT, Inmarsat, TELESAT, RCA SATCOM I and II, INTELSAT IV and V, WESTAR, and NAVSTAR (GPS). Each system's purpose varies, but in general these satellite based systems possess the following characteristics:

- Systems such as MARISAT and Inmarsat offer reliable, rapid, real-time, direct, high-quality service
- Communications coverage can be provided throughout the world. Such services available include; data transmission, teletype message services, facsimile, weather, safety related communications, and distress alerting (EPIRB)
- Accurate global navigation is provided by the DoD's navigation system, NAVSTAR GPS. This system consists of a collection of satellites which provides Universal Coordinated Time (UTC time) with an accuracy level on the order of microseconds and positioning information for terrestrial, maritime, or airborne platforms equipped with the appropriate GPS equipment accurate to within 10 meters

- Shipboard/shore installation and usage of satellite equipment is highly feasible as demonstrations have indicated with commercial shipping

The use of satellite based systems do, however, feature distinct disadvantages. Some of these include:

- Orbit utilization and frequency band allocation is an ongoing problem. Due to the substantial terrestrial use of the microwave frequencies, careful frequency planning is required. Frequency allocations often saturate large areas and prevent the implementation of certain satellite frequencies
- Interference from existing terrestrial microwave radio relay links within the designated satellite bands. Locating the earth station satellite receiving antennas is a problem
- Selected bands of frequencies must deal with ionospheric scintillation and atmospheric absorption
- Interference between satellite networks when common frequency bands are used is a problem
- The close spacing of co-frequency satellites is of concern
- There are limited bandwidth and polarization problems
- There is a requirement to maintain a stable platform for proper antenna pointing
- The system is susceptible to shadowing by mountains and buildings

Coast Guard's applications for satellite communications may include emergency distress (EPIRB), AMVER, Ship/Shore long range, weather, Marine Broadcast, International Ice Patrol Broadcast, SAR Control, data transfer, and a nationwide paging and positioning system. The satellite navigation system, GPS, is highly applicable to the DoD, Coast Guard, and the maritime public. The evolution of satellite communications technology is driven by the following significant trends:

- Increasing the capacity of bandwidth-limited communication satellites by utilizing frequencies more than once. (polarization and multiple-exclusive spot beams)
- Orbit utilization, frequency reuse, frequency interleaving
- Upgrading of multiple access techniques, e.g., FDMA, TDMA, CDMA
- Processing transponders to improve error performance will be upgraded
- Polarization designs of spacecraft and earth station antennas

- Optical communications-Aluminum Gallium Arsenide (AlGaAs) lasers for up and down links
- Design of narrow beam width antennas

12.4 Very Small Aperture Terminal (VSAT)

For fixed earth-based applications, the advent of the so-called VSAT has been making the use of satellite communications an attractive and economical alternative to the use of public switched telephone systems. VSATs are low cost satellite terminals which operate in the "Ku" and "C" frequency bands. They feature a relatively small antenna aperture (hence its name) typically 0.5 to 2.5 meters in diameter. VSAT systems generally consist of a large number of fixed remote terrestrial VSAT stations under control of a powerful master terrestrial station called a "hub". Normally, a centralized network topology, as illustrated in Figure 12-1, is employed for VSAT systems.

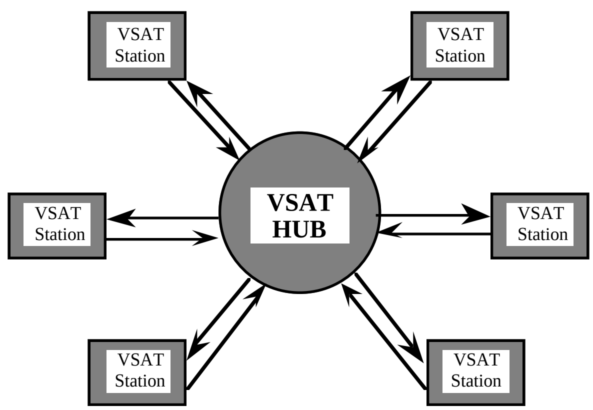
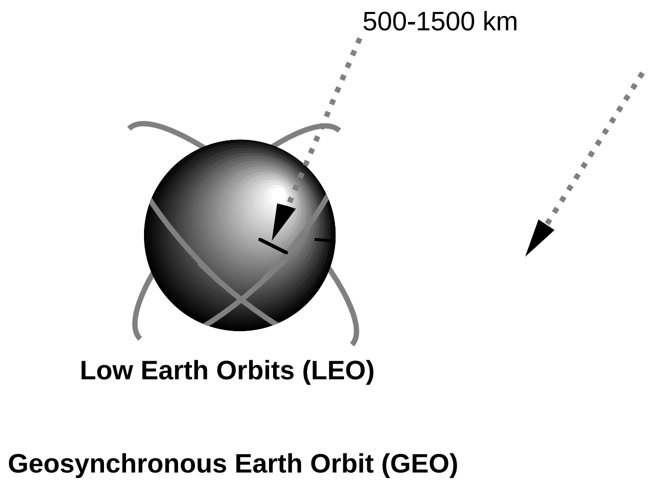


Figure 12-1. VSAT Network Topology

Typically, the hub transmits to remote VSAT stations, continuous bit streams which are accessible by all VSAT stations within the system. The bit stream consists of a sequence of delineated message packets each indicating the identity of the destination VSAT station. Transmissions from the VSAT stations to the hub generally employ some type of TDMA scheme. VSAT systems are normally implemented for applications in which a large number, ranging from hundreds to thousands, of remote stations need to send short messages according to a bursty traffic generation profile to a centralized facility. Typical data rates from VSATs to the central hub currently range from 600 bps to 64 Kbps. From a hub to the remote VSAT stations significantly higher data rates have been implemented. They typically exceed rates higher than 64 Kbps.

12.5 Low Earth Orbiting (LEO) Satellites

Although the 24 hour geostationary is the most commonly used orbit for communications satellites, one of the more significant developments recently in the area of satellite communications has been the emergence of small LEO satellite technology. Unlike Geostationary Orbiting Satellites (GEOs) which feature an orbital altitude of 36,000 km, LEO satellites circle the earth at altitudes between 500 -1500 km.



13 COMPUTER SYSTEM ARCHITECTURE CONCEPTS

13.1 Overview

This section provides a brief overview of selected topics of current significance in the area of communications system development. The topics chosen have been those technology areas associated with the so-called “open systems” architecture philosophy. For the Coast Guard, the significance of this architectural philosophy is the design and implementation of communications systems which can maintain their technological superiority after deployment/implementation. The use of commercial-off-the-shelf “open” products, both hardware and software, as well as the custom development of hardware and software, meeting this “open” criteria permits maximum technology insertion to be achieved without significantly impacting logistical problems. Implementing this "open" concept provides to the Coast Guard the following benefits:

- Continued technical superiority through rapid technology insertion
- Lowered cost through:
  - Competitive sources for the hardware/software components
  - Reusable components
- Logistical simplification through component modularity and commonalty

13.2 Open System Interconnect (OSI)

Integration of communicating computing systems has become more and complex as these individual systems have themselves become more complex and diverse. To accommodate the communications requirements amongst these many diverse computing platforms, the International Standards Organization (ISO) developed the so-called OSI seven-layer model. Figure 13-1 illustrates and summarizes the seven-layer OSI reference model.

LAYER		FUNCTION
7	Application	Selects appropriate service for applications
6	Presentation	Provides data conversion/ reformatting
5	Session	Coordinates end-to-end interaction
4	Transport	Provides end-to-end integrity of data & quality of service
3	Network	Switches and routes data through the underlying network
2	Data Link	Transports blocks of bits across physical link
1	Physical	Transmits information across physical media

Figure 13-1. ISO Open Systems Interconnect Reference Model

This was developed to foster open communications among a wide variety of heterogeneous products freeing implementors from concern with any proprietary communications solutions. They goal of OSI is to make these products interoperable with any other product in the communications network. The hoped for by-products of OSI standardization has been to drive down product costs, improve the overall quality of these products, and to foster the expansion of the communications domains into new diverse geographical areas. ISO is an international standards body. Its membership is drawn from various national organizations such as the ANSI in the United States and the British Standards Institute in the United Kingdom. ISO and the CCITT maintain a close relationship in developing international communications standards.

OSI is a reference model, a collection of service definitions, and a collection of protocol specifications. The service definition for each layer defines the services to be provided to the user of that layer – i.e., the layer above it. A brief synopsis of each OSI layer is provided below:

- Physical Layer. The physical layer provides the electrical encoding and decoding of information for transmission over the physical media between directly "connected" communications systems. The media ranges from copper and fiber optic cables to RF propagation pathways. This layer is responsible for specifying the manner in which the physical circuit is to be activated, maintained, and deactivated.
- Data Link Layer. The data link layer manages and provides access to the physical link between two communications systems. In local area networks, the functionality of the data link layer is separated into two components – the LLC and the MAC. Here, the LLC is responsible for placing data into transmission frames, addressing the frames to its intended destination, performing error checking on the binary data, and applying any necessary flow control. MAC is responsible for arbitrating multiple system access to the physical media.
- Network Layer. The network layer provides the routing and relaying functions between the communicating peer systems separated by a single subnetwork or through multiple subnetworks. These subnetworks include both LANs and Wide Area Networks (WANs).
- Transport Layer. The transport layer is responsible for providing a reliable error-free end-to-end transfer service to the service user. These services include (1) data sequencing (2) end-to-end flow control (3) multiplexing (4) error recovery.

- Session Layer. The session layer provides a set of tools for its users. These tools provide for the orderly (synchronized) transfer of data and control information between peer open systems.
- Presentation Layer. The presentation layer deals with the "presentation" of data. Whereas the Application layer above is concerned with the semantically nature of information, the presentation layer provides the syntactical description of the information. For instance, an integer can be described in many different ways – binary, ones-complement, twos-complement, binary coded decimal, etc. These are all different "presentations" of the same piece of information. The presentation layer allows peer systems to negotiate and translate these presentations to allow dissimilar systems to communicate.
- Application Layer. The application layer is that portion of OSI most visible to the end user. It provides services to the user communications application. Some common OSI application layer "functions" include:
  - File Transfer, Access, and Management (FTAM)
  - Message Handling Systems (MHS), better known as X.400
  - Virtual Terminal (VT)

13.3 Government Open System Interconnection Profile (GOSIP)

The most rapidly growing sector of high technology industry in the United States today has been unquestionably the use of telecommunications links for data transmission. There has been an ongoing need for a set of standards to ensure that end-to-end communications systems are compatible with each other as far as their ability to exchange information is concerned. Any incompatibility implies the need for time-consuming and costly conversion processing. In the past, U.S. Governmental agencies had difficulties sharing information due to any common networking standards. This condition became more exasperated by the rapid recent advances in communications technology. Thus, in order to implement and enforce common networking standards, the so-called Government Open Systems Interconnection Profile (GOSIP) was born. GOSIP is meant to provide many benefits to the U.S. Government. These include:

- Cost reduction – reducing software development costs for communications functions, reducing training costs as personnel become efficient with **standard** technology, stimulating a competitive communications product marketplace
- Increased communications capability through interoperability

As the name implies, the Government embraces the ISO OSI standard. The premise upon which OSI technology was embraced by the Government stem from its belief that the inevitable consequence of OSI technology will be smaller, less costly, and higher performance computing systems. To support the strongly held belief that Federal Agencies can benefit greatly from OSI technology, GOSIP provides a technical specification with sufficient detail for Federal agencies to acquire and effectively use OSI products. It is the vehicle through which the process of assimilating OSI technology into Federal agencies can be performed with ease.

GOSIP defines and describes a common set of communications protocols which enable systems developed by disparate vendors to interoperate and seamlessly communicate. These protocols are developed by either the ISO or the CCITT. The so-called GOSIP stack is based upon agreements reached by active participants (vendors and users of computer networks) in the National Institute of Standards and Technology (NIST) workshop for implementors of Open Systems Interconnection.

The original version of GOSIP (version 1) specified only basic OSI protocol functionality. It featured:

- X.400 Message Handling System (MHS) (circa 1984)
- File Transfer, Access, and Management (FTAM)
- ISO 8802-3 (IEEE 802.3 – Ethernet)
- ISO 8802-4 (IEEE 802.4 – Token Bus)
- ISO 8802-5 (IEEE 802.5 – Token Ring)

The current version of GOSIP (Version 2) was mandated by the Government in October 1992 and features the following additions to GOSIP version 1:

- Integrated Services Digital Network (ISDN)
- Office Document Architecture (ODA)
- Virtual Terminal (VT)
- End System-to-Intermediate System (ES-IS) Network Protocol
- Connectionless Transport Service (CLNP)
- Connection-Oriented Network Service (CONS)

Figure 13-2 illustrates the GOSIP Version Protocol Architecture:

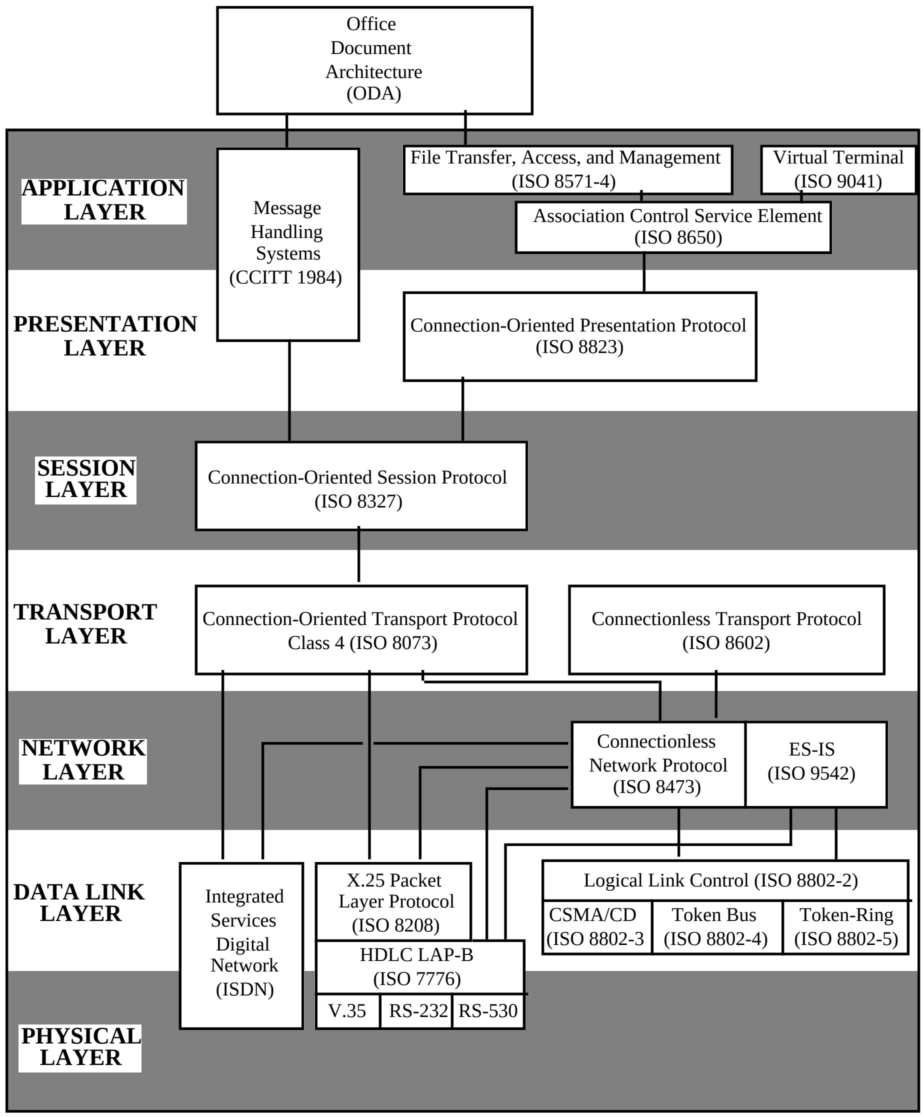


Figure 13-2. GOSIP Version 2 Protocol Architecture

Currently, the acceptance of GOSIP has been hindered by the lack of critical protocol components. This problem has been addressed by the work currently proceeding within the NIST Open Systems Interconnection workshop on GOSIP version 3. The enhancements with this version of GOSIP will include:

- X.500 Directory Services
- X.400 Message Handling System (MHS) (circa 1988)
- Fiber Distributed Data Interface (FDDI)
- Transaction Processing

- Manufacturing Message Service
- Remote Database Access
- Electronic Data Exchange
- X-Window System
- Virtual Terminal (VT) Extensions
- File Transfer, Access, and Management (FTAM) Extensions
- Transport Protocol Class 2 (TP2)
- Information Retrieval

It is widely assumed by industry experts and federal users that GOSIP compliance in the Federal Government will accelerate once GOSIP version 3 becomes mandatory sometime in 1994.

13.4 VMEbus

The blossoming popularity of the VMEbus in the commercial marketplace has created a large vendor base, offering a wide range of high performance processor boards for use in real-time applications. Furthermore, the current snapshot in time shows the VMEbus as by far the leading architecture being favored by the Army, Navy, and Air Force. It is the preferred Commercial-Off-the-Shelf (COTS) option for designers of high performance systems Many of these programs will continue into the latter part of this decade. Furthermore, the Navy has unequivocally demanded that ALL future systems be based on Futurebus+ and that current implementations be compatible with Futurebus+ upgrades. The VMEbus has been consistent with this Navy mandate due to its integral bridge specification.

For Coast Guard development efforts, adopting the VMEbus open-bus architecture offers the benefit of access to this wide and readily available collection of COTS computing elements. These components can be integrated in a modular and expandable fashion into high performance processing engines. Furthermore, there exists a prolific amount of VMEbus interface cards to other buses (e.g., MIL-STD-1553, Navy Tactical Data System (NTDS), FDDI, SCSI). In addition, there exists another distinct advantage to using the VMEbus as the standard platform for systems as it can be tested and verified with low cost commercial hardware.

The VMEbus has two 96-pin DIN 41612 and DIN 41494 type connectors on a 233.35 mm by 160 mm "Eurocard". It consists functionally of the following four components:

- Data Transfer. The VMEbus permits data to be passed 8, 16, or 32 bits in parallel and supports address paths of 16, 24, or 32 bits. This makes it possible for 8, 16, and 32 bit processors to share the bus.
- Arbitration. The VMEbus uses centralized arbitration to the extent that a single board acts as a global arbitrator for four arbitration lines. Boards may be designed to "release-when-done" (e.g., DMA controllers) or "release-on-request". The latter implementation is generally used in order to allow high priority bus users to preempt lower bus users. The VMEbus allows a myriad of arbitration schemes ranging from simple round-robin to priority arbitration.
- Interrupt Handling. Seven prioritized interrupt lines are provided by the VMEbus. In response to an interrupt, the interrupter also passes an 8 bit interrupt identifier to the interrupt handler. This means that 7*256 different interrupt status numbers can exist in a VMEbus system.
- General Bus Utilities. The VMEbus provides the following utility signals on the bus which provides (1) a 16 MHz clock signal (2) 5V, -12V, +12V supply lines (3) a fail line which can be asserted by a board to generate a signal to indicate failure (4) system reset line (5) an alternating current (AC) fail line giving boards in the system at least 4 ms of warning that direct current (DC) power is going outside its specified limits.

Figure 13-3 illustrates a VMEbus board.

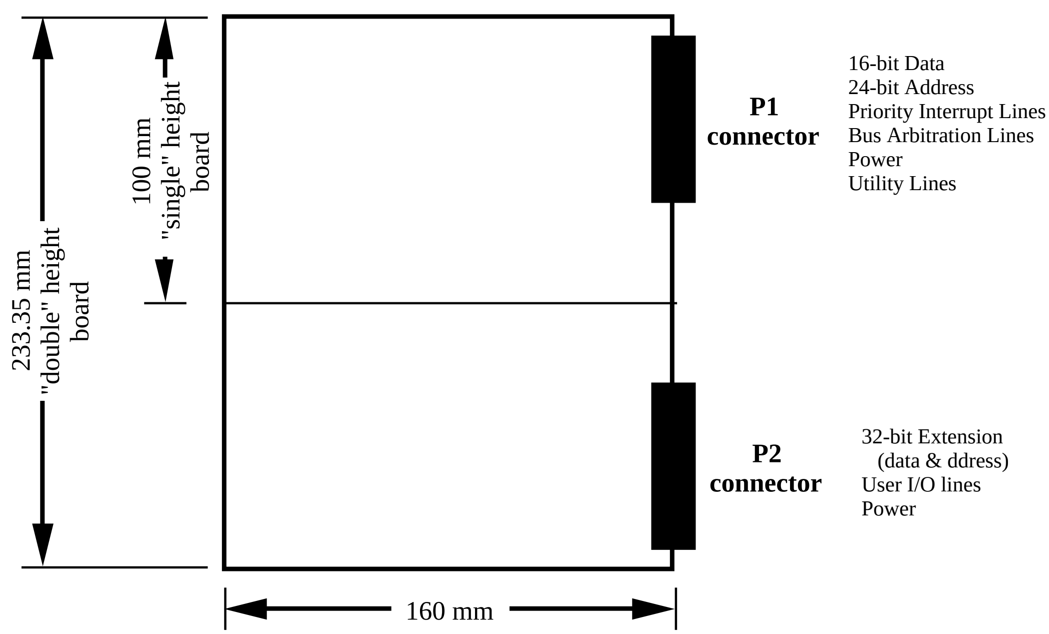


Figure 13-3. Standard VMEbus Board

The current trend in VMEbus technology has been towards providing more functionality on a single VMEbus board. For instance, current technology permits the placement of multiple microprocessor, SCSI, multiple serial (RS-232/422), and megabytes of memory on a single standard VMEbus board. Furthermore, there have been some recent developments which extends VMEbus performance into the midrange level of Futurebus+ performance. A new Source Synchronized Block Transfer (SSBLT) mode for VMEbus has been defined. There is also a new emerging 64-bit VME standard called VME64. In this new standard, the 32 data and 32 address lines of standard VME are multiplexed. Thus, VME64 offers 64-bit data transfers and 64-bit addressing. SSBLT is to be implemented under VME64 and it is estimated that under this new VMEbus standard bus transfer rates will exceed 160 megabytes/second, the current midrange of Futurebus+ performance.

13.5 Futurebus+

The development of Futurebus+ has been the product of the joint U.S. NAVY and industry work group, the Next Generation Computer Resources (NGCR) program and IEEE, respectively. This standard is anticipated to be the basis for rapid new future developments advancing the state of the art in processor, memory, and I/O designs. Futurebus+, more so than with the VMEbus, will permit system developers to focus on issues other than how to talk between dissimilar boards from different vendors.

The advantages of Futurebus+ over the VMEbus are in throughput and flexibility. Futurebus+ throughput is of the order of a few hundred megabytes/second based on current component technology, typically four times the maximum standard VMEbus bus transfer rate. This advantage for the future Coast Guard systems translates to less bus latency and shorter data transfer queue times. Another advantage of Futurebus+ is that it provides a means to replace modules without removing power from the module ("hot extraction"). Futurebus+ does not use daisy chained data or arbitration signals like the VMEbus.

The hardware design of Futurebus+ accommodates the best qualities of current design practice. It also incorporates features most appropriate for future designs. Futurebus+ is not modeled after any vendor specific Central Processing Unit (CPU) control signal pattern but, rather, is an open architecture capable of efficient use by many manufacturers. Even the newest massively parallel and data flow designs can be interfaced into the multiple master bus environment. Since Futurebus+ defines a standard method of communications between peers, any application whose requirements can be met within the aggregate bandwidth of the bus can be designed using Futurebus+. Futurebus+ does not limit the selection of CPU or the mixture of widely differing CPUs in the same system.

Although there currently exist some Futurebus+ products in the commercial marketplace, we are still in that initial technological ramp-up time for Futurebus+ vendors, preceding the start of the product cycle where the COTS benefits accrue (i.e., cost, large product line, large vendor base). In actuality, Futurebus+ is still undergoing refinements. For the near term, for Coast Guard systems being designed with VMEbus with an eye toward Futurebus+ should be aware and in tune with those ongoing efforts with respect to the VMEbus' so-called Futurebus+ "bridge".

14 SECURITY

14.1 Overview

The techniques and technology associated with computer systems security , their interconnecting communications systems, and the information processed and transferred all constitute elements of information security (INFOSEC). This is a key component of any communications technology. The scope of INFOSEC in the commercial world is already very well documented and exceeds the scope of this document. Consequently, this section highlights some of the important security issues of which present Coast Guard communications system implementors should be aware. In particular, it focuses in on those security issues confronting those with a requirement to use or interact with U.S. Navy communications assets. For historical information regarding the evolution of INFOSEC as well as the survey of specific commercial based INFOSEC technology and techniques, the reader is directed to the voluminous collection of information regarding this subject matter.

Security requirements affect both hardware and software implementations in an INFOSEC environment. Not only do Computer Security (COMPUSEC), Communications Security (COMSEC), Transmission Security (TRANSEC), and Network Security (NETSEC) requirements apply, but also Operational Security (OPSEC). The application of security policies and standards must also be considered in the overall communications system design. The security technology task is integral to the overall system engineering efforts, from requirements allocation and performance assessment through implementation and system certification. New Coast Guard systems with interoperability requirements with Navy assets must address security issues up front and in great depth. The defined security policy must be reflected in any system level specification. Henceforth, any remaining security issues should focus exclusively on the implementation and accreditation in the time frame and cost required by the Coast Guard system milestones. The following is an overview of these applicable security technologies including their impact upon any new and applicable Coast Guard communications system development.

14.2 COMSEC Technology

COMSEC encompasses link level techniques to secure communications by encryption, TRANSEC for transmission over public airwaves, and NETSEC for routing and control within networks. Commonly, security services are provided by cryptographic hardware and include encryption/decryption, authentication, anti-spoof, and rekey. Modern cryptographic equipment also provide key unwrap, and, increasingly, key generation. In addition, Traffic Flow Security (TFS) might be imposed for links which pass critical information at infrequent intervals. These security services are implemented with equipment ranging from stand-alone COMSEC cryptos such as the KG-84 and KY-57 to embedded COMSEC cryptos such as the OVERTAKE family of embeddable chips/modules. For these equipment/devices, performance characteristics must be determined and risks associated with development/endorsement evaluated.

Several new products have been in development under the Commercial COMSEC Endorsement Program (CCEP), User Partnership Program (UPP), and co-operative Memorandum of Agreements (MOAs). They have been extensively examined by the Navy through their Space and Naval Warfare Systems Command (SPAWAR) and Naval Research Laboratory (NRL) to determine the ability to collectively leverage from in-process developments for risk reduction in support of new Navy initiatives. These products include WINDSTER, INDICTOR, TEPACHE, and FORESEE. The primary issue has been the incorporation of these technologies into the a modular functional package. This has been referred to, in general, as the Modular Security Device (MSD).

Critical aspects of any COMSEC-based approach are the monitor, control, and key management/distribution mechanisms. These considerations can impact operational costs and the operational feasibility of a distributed COMSEC approach. Developments such as the Single Point Keying (SPK) concept currently being developed by the National Security Agency (NSA) could affect COMSEC design options, if implemented in a networking system architecture. The SPK concept allows remote electronic key fill from a single point on an afloat platform. Furthermore, it is compatible with NSA's Electronic Generation and Distribution key management concept. To support COMSEC operations, a system Key Management Plan (KMP) must be developed and approved prior to system certification. The KMP normally describes methods to allow key distribution using existing keying equipment such as KOI-18, KYK-13, and HYX-15 or through future key fill equipment.

The primary factor affecting cost and schedule in implementing embedded encryption capabilities successfully in new communications system is at the level of security integration with the target environment. In spite of the high level of integration of security critical functions in CCEP technology, experience has shown that several years are generally required to complete the design evaluation and certification process.

14.3 COMPUSEC Technology

The primary shortcoming of traditional COMPUSEC for Navy systems has been the inability to support Multi-Level Security (MLS) communications. COMPUSEC technology has evolved rapidly over the last few years, with most work focused on the development of MLS operating systems. Several available MLS operating systems have been approved by Designated Accreditation Authorities (DAAs) for B1 levels; several other options have been accepted to enter certification cycles for B2 level. Initiatives are also underway to develop A1 level MLS operating systems. Only recently, in systems such as the SUN Secure UNIX development effort, have communications capabilities become integrated with the underlying operating system. This still does not provide support for interprocess communications between MLS application processes, such as system protocol functions. There are several current development efforts that address this critical area. It is reasonable to anticipate that this shortfall will be resolved in the near-term as more emphasis is placed on fully integrated and interoperable tactical communications systems requiring computer control. There is indeed evidence that COMPUSEC technology is maturing to a point where we may soon see significant benefits, in terms of hardware cost reductions, reduced form factors, and less development risk in maintaining approval authority within the U.S. Navy. COMPUSEC also is maturing in its ability to be fully integrated with INFOSEC, permitting complementary security functions. To the degree that COMPUSEC is incorporated (in components or security filters for bypass channels), certification will require a system-level approach and interpretation of requirements beyond the operating system view of COMPUSEC.

14.4 Management of Certification/Accreditation Process Risk

The certification and accreditation of security must be viewed from an overall system level, including development, operational and life-cycle support considerations. The key to quantitative assessment of an architecture is the security baseline, including the Threat and Vulnerability Analysis (T/VA) and Certification Requirements Document. COMPUSEC certification requirements in the "Yellow Book" are particularly sensitive to the threat environment of security levels processed and security levels of system users. The B3 level criteria would apply to most COMPUSEC alternatives, however implementation on this level could restrict future system evolution. A combined use of COMSEC and B2 level COMPUSEC is probably the most attractive alternative to be considered. COMPUSEC certification evaluation provided by NSA (if requested), is advisory to the Designated Accreditation Authority. For COMSEC technology, NSA has the charter to approve the application of existing products within a system, development of new products, and embedding of COMSEC in other system components. The level of risk can be mitigated by the use of approaches based on the use of existing approved products or components. It is likely that a combination of embedded COMSEC and COMPUSEC will be appropriate for future service communications systems. To arrive at an acceptable level of risk, it will be necessary to pursue parallel development of these two critical security techniques.

14.5 U.S. Navy Security Background

Current Navy user systems operate in a system high mode at security levels from Secret to Top Secret with multiple compartments. Currently, links are not shared and COMSEC/NETSEC operations are provided by link encryption. TRANSEC typically is furnished by link controller equipment. On current afloat platforms, minimum access is at the Secret level, limiting the potential threats and vulnerabilities the system must counter. Although this "RED ship" concept limits the range of access levels, the requirements of future systems operating in open environments with a wider spectrum of security threats is now being seriously considered.

The Navy system provides COMPUSEC, COMSEC, TRANSEC, and NETSEC capabilities to protect user and network control data within the platform environment and over communications media. As the Navy has gone toward integrating the communications assets of its individual services into a collective framework, a wide range of architecture has been proposed and evaluated to satisfy the overall Navy mission requirements. The factors involving security architecture evaluation are summarized in Figure 14-1.

Security Architecture Evaluation Factors

- User and Network Protection / Failure Modes
- Performance Factors
    - Throughput
    - Delay
    - Efficiency (Overhead, internetworking support)
- Availability (fault tolerance, reconfiguration)
- Transition Support
- Compatibility with Existing Systems
- Incremental Addition of New Systems
- Flexibility - Support for Evolving Requirements
- Depth - Sensitivity to Platform Environment
- Technical Risk / Maturity of Technology
- Certification Process Risk (History)
- Control (Number of Organizations Involved)

Figure 14-1. U.S. Navy Security Architecture Evaluation Criteria

The most difficult factors to evaluate are those relating to technology maturity and certification/accreditation process risk. Because these are subjective processes, the assessments must be based on evaluation of previous experience rather than theoretical considerations.

15 COMMUNICATIONS AND NETWORK SERVICES

15.1 Global Maritime Distress and Safety System (GMDSS)

The Global Maritime Distress and Safety System or GMDSS is a new radio communications system for larger commercial ships. Cargo ships of 300 gross tonnage in international waters, and passenger ships carrying more than 12 passengers in international waters are effected. Ultimately it will replace the current ship-to-ship safety system which relies on a manual Morse code system on 500 kHz and voice radio telephony on Channel 16 and 2182 kHz, with an automated ship-to-shore system using satellites and digital technology.

The GMDSS requirements do not apply to fishing, recreational, or small passenger ships operated on domestic voyages. Eventually the GMDSS will replace the current safety system throughout the world; consequently, these ships may ultimately be affected by the changes brought about by full implementation of GMDSS.

The GMDSS changes are mandated by international treaty obligations. In 1988, the International Maritime Organization (IMO), an organization of the United Nations, amended the Safety Of Life At Sea (SOLAS) Convention to implement the GMDSS worldwide. The United States has been a strong advocate of the GMDSS internationally. The FCC adopted the GMDSS regulations for all U. S. compulsory ships as of January, 1992. Advantages of the GMDSS to the old system are numerous. They:

- Provide worldwide ship-to-shore reporting and does not rely on passing ships
- Simplify radio operations, alerts may be sent by a simple "push of a button"
- Ensure redundancy for communications; it requires two separate system for alerting
- Enhance search and rescue, operations are coordinated from centers located ashore
- Minimize unanticipated emergencies at sea, maritime safety broadcasts are included in normal communications
- Eliminate reliance on a single person for communications, it requires at least two licensed GMDSS radio operators and typically two maintenance methods to ensure distress communications capability at all times
- Automate watch keeping; it eliminates the need for manual watch keeping on the Morse code frequency and voice channels. Replaces these with selective calling on the Digital Selective Calling (DSC) circuits or satellites

The implementation process will be a phased approach. New construction ship will require GMDSS by 1995 and full phase in will be completed by 1999, at which time the old “radiotelegraphy” will no longer be recognize as a distress call circuit. Ship desiring to carry both systems may do so and simply monitor GMDSS, however, certification under GMDSS will require full compliance with GMDSS regulations.

In general terms all GMDSS equipped ships must carry a 406 MHz EPIRB, a VHF radio capable of transmitting and receiving DSC and radiotelephony, a NAVTEX receiver, a 9 GHz Search and Rescue Transponder (SART), and two-way VHF portable radios. Ships must also have a dedicated, non-scanning DSC watch receiver for VHF Channel 70 (156.525 MHz), and continue to carry until 1999 the current watch receiver/alarm generator installation for distress frequency 2182 kHz. Most GMDSS ships must also carry MF-DSC radio equipment, either HF radios (DSC and NBDP) or Inmarsat ship earth stations, additional DSC watch receivers and facilities for marine safety information, depending on the intended voyage of the ship. See FCC GMDSS regulations for exact details. During the phase in period current equipment with appropriate capabilities may be employed, but in 1999, and for certification, all equipment must be FCC approved for GMDSS use.

15.2 U.S. Coast Guard Communications Networks

The Coast Guard Data Network (CGDN) provides the primary conduit for day to day communications between most land based Coast Guard units nation wide. The network consists of both Coast Guard owned facilities and leased circuits. The network has been known under several different names and configurations. Currently the Coast Guard is transitioning to FTS2000 as their primary carrier of electronic and voice messages.

The CGDN is an X.25 packet switching data communications network. The protocol is efficient and cost effective for the volume of traffic normally encountered, however this protocol is not efficient for very large messages. Packet switching lends itself favorably to small messages and fits the Coast Guard needs in most cases.

The backbone of the network is made up of an interconnection link between major nodes at a rate of 56 Kbps. This backbone network has an architecture to enable re-routing of message traffic in case of outages or emergencies. The major nodes are further connected to secondary nodes, normally running at 9.6 Kbps, but may run at 56 Kbps, depending on traffic volume and requirements.

In areas where traffic volume does not warrant an “on line” system access a dial-up format is used. The user dials a active node to retrieve his messages. The major disadvantage of a dial up system is that the messages must be stored until retrieved by the user, and the inherent delays incurred with this type of system.

Network security is afforded by implementation of the Digital Encryption Standard (DES), but this system does not clear the network for transmission of classified material.

The CGDN is currently being interfaced with the CG Standard Work Station (CGSWS) at most major shore nodes. Plans are underway to include connections at many port facilities to enable CG Cutters access to the network. In addition for areas not served by the CGDN, a LAN interface will be implemented.

The Secure Data Network (SDN) is designed to replace the Coast Guard's Secure Command and Control Network (SCCN). Currently the primary security device for the low volume secure network is the STU-III interface with the CGSWS. An "Electronic Link Filter" developed by the Coast Guard Information Systems Center (ISC) has enabled the STU-II and SWS to work together.

The CGDN plans are under consideration to extend this capability to port cities for cutter access.

15.3 Automatic Digital Network (AUTODIN)

AUTODIN is managed by Defense Information Services Agency (DISA), for the DoD. Coast Guard has limited access at selected locations. AUTODIN is a primary conduit for record copy message traffic. It affords full security in accordance with the NSA and is capable of high volume, prioritized message traffic. The vast majority of Coast Guard classified message traffic is transferred over AUTODIN.

15.4 Global Positioning System (GPS)

The DoD-implemented GPS provides "real-time" position fixing capabilities for earth-based, marine, and airborne platforms. Aside from three dimensional position fixing, the GPS system provides users with velocity information, the coordinated universal time (UTC). It is the most reliable global navigation system available today. It will most likely be the accepted universal means for navigation positioning well into the next century. It features a rough accuracy of about 25 meters, although implementation specific error sources decrease this accuracy.

The GPS satellite system consists of a constellation of satellites in six circular orbits with identical periods of 12 sidereal hours. The satellites circle the earth at an altitude of 20,200 km. The orbits are equally spaced with inter-plane angle of 120°. The inclination of the orbital planes (angle with equatorial plane) is 55°. The fully operational configuration consists of 18 satellites equally allocated to each of the circular orbits (3 per orbit). In order to provide users at any point near the earth's surface with continuous navigational information from four satellites, satellites of consecutive planes are separated by 40°. In addition, after a satellite traverses across the equator northward, will pass the equator in the other southward direction.

The underlying manner in which a user in a GPS system determines its three dimensional position is based on Cartesian geometry. The unknown user position is (x_u, y_u, z_u). The "known" positions of four GPS satellites are, respectively, (x_1, y_1, z_1), (x_2, y_2, z_2), (x_3, y_3, z_3), and (x_4 ,y_4, z_4).

If R_1, R_2, R_3, and R_4 denote the distance from the user to each of the satellites, then, according to Cartesian geometry:

$$(x_1-x_u)^2 + (y_1-y_u)^2 + (z_1-z_u)^2 = (R_1)^2$$

$$(x_2-x_u)^2 + (y_2-y_u)^2 + (z_2-z_u)^2 = (R_2)^2$$

$$(x_3-x_u)^2 + (y_3-y_u)^2 + (z3-z_u)^2 = (R_3)^2$$

R_1, R_2, and R_3 are computed based on synchronized docks within the GPS system. Knowing the velocity "c" of the signals from the satellites (300,000 km per second), the R values are computed as:

$\mathbf{R} = (c)\Delta t$ where Δt = time difference from signal generation (satellite) to receipt (user)

If the user and satellite docks were always perfectly synchronized, then establishing a user location would only require three satellites (three equations, three unknowns). However, the user clock will have same unknown time bias (i.e., Δt = actual delta time plus an unknown bias factor—3 equations, 4 unknowns). This necessitates the use of the fourth satellite.

Nevertheless, the predicted position of a GPS satellite will inevitably contain slight errors in each of the three dimensions causing errors in the computed position of an user. *Differential* GPS minimizes these satellite positional errors and thereby improves the overall accuracy of GPS down to an order of just a few meters. The basic premise employed by *Differential* GPS is that a "second" user at a "significant" terrestrial distance from the first, for instance a 200 km separation, will experience the same errors. From an overall earth-space perspective, 200 km is effectively negligible compared to the 20,200 km height of the GPS satellite. Consequently, the signal paths from the satellite to both users can be seen to be identical. Their paths would nearly coincide and would, for example, experience the same ionospheric refraction. In *Differential* GPS system, these "second" users are stationary shore based locations with precisely known x, y, z coordinates – i.e., latitude, longitude, and height. The errors in the position computed stationary user can easily be determined and be applied to the user seeking actual positioning information. The technique whereby this additional "differential" error information is used by the user, forms the crux of *Differential* GPS. GPS receivers being used as navigational devices receive GPS satellite range corrections for a fixed shore receiver whose antenna position is precisely known. The navigational GPS receivers adjust their own measured satellite ranges based on information received from the fixed shore receiver to a more accurate range value. This adjusted range is then used for the navigation computations performed by the remote navigation device. For a more technically substantive treatise on this subject, the reader is directed to *The Application of NAVISTAR Differential GPS in the Civilian Community*.

15.5 International Maritime Satellite (Inmarsat)

The Inmarsat began operations in 1982. It has been patterned after INTELSAT as a commercially oriented international cooperative organization. It's purpose is to provide mobile satellite communications to the international maritime and offshore industries. This maritime market includes offshore drilling rigs and production platforms, general cargo and container ships, large yachts and passenger vessels, and small coastal vessels such as fishing vessels and pleasure boats. Services provided by Inmarsat include telephone, facsimile, and telex, as well as leased lines for voice and high speed data communications. The Inmarsat consists of three components. These are:

- Coast Earth Stations (CESs)
- Space segment
- Ship Earth Stations (SESs)

The space segment consists of geostationary satellites positioned over the Atlantic, Pacific, and Indian oceans. They provide communication links with mobile vessels in nearly all navigable ocean areas. Worldwide coverage over both land and sea is provided by satellites in geostationary orbits above the equator at an approximate height of 36,000 km at longitudes of 180° E (Pacific Ocean), 55.0° W (West Atlantic Ocean), 18.5° W (East Atlantic Ocean), and 63° E (Indian Ocean).

The CESs are large ground facilities. They provide the link between the satellites and telecommunication networks ashore (e.g., landline connections to public switched telephone systems). Communication between the satellites and the CES occur in the C-band. Transmission from the CESs to satellite occurs at 6 GHz and reception occurs at 4 GHz. A typical CES consists of a parabolic antenna 15 meters in diameter for C-band transmissions accompanied by another dedicated antenna for L-band network control transmissions.

The SESs are SATCOM terminals onboard vessels. Communication between the SESs and satellites occur in the L-band at 1530-1545 MHz (satellite-to-SES) and 1626.5-1646.5 MHz (SES-to-satellite). Currently, two types of SES are employed in the lNMARSAT system. These designated respectively as the Standard-A and Standard-C SES.

The Standard-A SES uses a 1 meter diameter stabilized parabolic antenna. It provides capabilities for telephone, facsimile, data communications, and telex via a TDMA access scheme. Data rates may be up to 2400 bps for voice band data and as high as 56 Kbps for ship-to-shore high speed data communications. Standard-A SESs provide a distress priority capability whereby a telephone or telex channel is automatically allocated via a CES to a rescue coordination system.

The Standard-C SES has been introduced to provide telex and low speed data communications in the smallest feasible physical package. The maximum possible data transmission rates supported by Standard-C is on the order of hundreds of bits per second. The intent of the Standard-C designation was to provide an Inmarsat capability to the smallest of vessels down to lifeboats. The antennas for Standard-C SESs can be as small as 15 cm and can be mounted anywhere on top of a vessel without stabilization

The implementation of Standard-C requires careful analysis, design, and implementation to the limitations imposed by L-band communications. These limitations include:

- Phase noise for low data rates
- Small antenna gain from characteristic broad beam unstabilized antenna
- Multipath fading from signal reflection off sea surface
- Doppler offset from ship motion
- Shadowing effects from vessel's superstructure

16 FLEET SATELLITE COMMUNICATIONS (FLTSATCOM)

16 FLEET SATELLITE COMMUNICATIONS (FLTSATCOM)

16.1 Overview

The U. S. Navy Satellite Communications System provides communications via satellites between designated mobile units and shore sites. These links supply worldwide coverage between the latitudes of 70 degrees north and 70 degrees south. Three satellites constellations are currently in use: Gap filler, LEASAT, and FLTSATCOM. The next generation satellite will be UHF Follow-On (UFO) which will provide increased capacity of current FLTSATCOM satellites. The SATCOM system includes satellites, RF terminals, subscriber subsystems, personnel, training, documentation, and logistic support.

Installation in support of the system are located on ships, submarines, mobile vans, aircraft, and at shore stations. These installations vary in size and complexity, depending upon the communications requirements of each location. The system has various processors which control RF links for message traffic and voice communications.

Although any part of the system may be operated as a separate entity, the integrated system provides connection for message traffic and voice communications to DoD long-haul communications networks. Certain shore stations provide a backup capability to other shore stations if an outage occurs. This backup capability is constrained to selected subsystems and to those shore stations that have the ability to access various satellites.

16.2 Satellites

Currently, three satellites are used by the Navy for UHF SATCOM. These are the Gapfiller, FLTSAT, and LEASAT. By the middle of the 1990's, the UHF Follow-On (UFO) satellites will be phased into service for UHF SATCOM.

Gapfiller is the name given by the Navy to the MARISAT satellites leased by the Navy from the COMSAT General Corporation. The Navy has leased the UHF section of these satellites which consist of one 500 kHz and two 25 kHz channels. Communications on the wide band channel are multiplexed through FDMA. This channel has been separated into subchannels with transmission data rates ranging from 75-2400 bps

The FLTSAT satellite contains an UHF, SHF, and an S-band communications antennas. Five operational FLTSAT satellites are currently deployed. Two of these satellites contain the FLTSAT EHF Package (FEP) which provides EHF capability for the tri-services. Each FLTSAT satellite has ten 25 kHz, twelve 5 kHz, and one 500 kHz channel. The 25 kHz channels have all been allocated for Navy use. All (23) channels have three different frequency plans for both the downlink and uplink to mitigate interference effects at those points on the earth's surface where satellite coverage overlaps. For those FLTSAT satellites with the FEP, an additional uplink and downlink frequency are operated. These frequencies are in the EHF frequency band at approximately 20 GHz for the downlink and approximately 44 GHz on the uplink.

The LEASAT satellite has seven 25 kHz UHF downlink channels and one SHF uplink channel. The UFO satellite system will consist of a constellation of satellites providing communications over four coverage areas – the continental United States, the Atlantic Ocean, the Pacific Ocean, and the Indian Ocean. The constellation will consist of two satellites in geosynchronous orbit over each coverage area.

16.3 UHF SATCOM

The Navy UHF SATCOM System is the primary component of the Navy's SATCOM system. It supports various information exchange user subsystems that use the satellites as relays for communications, control, and quality monitoring subsystems that provide data to manage satellite resources. Each user subsystem structure addresses specific naval communications requirements and are briefly described in Table 16-1.

Table 16-1. UHF SATCOM Major User Subsystem

SUBSYSTEM/USER	DESCRIPTION
Fleet Broadcast Subsystem (**FLTBCST**)	This is an expansion of the Fleet Broadcast, which historically has been the central communications medium for operating naval units.
Common User Digital Information Exchange Subsystem (**CUDIXS**)/ Naval Modular Automated Communications Subsystem (**NAVMACS**)	These two subsystems combine to form a communications network for transmitting general service message traffic between ships and shore terminals.
Submarine Satellite Information Exchange Subsystem (**SSIXS**)	The SSIXS complements other communications links between SSBN and SSN submarines and shore terminals.
Secure Voice Subsystem (**SECVOX**)	This is a narrow band UHF subsystem that links voice communications between ships and connects with wide area voice networks ashore.
Tactical Intelligence Subsystem (**TACINTEL**)	This subsystem is specifically designed for special intelligence communications.
Teletypewriter Subsystem (**TTY**)	This subsystem is an expansion of terrestrial teletypewriter transmission networks.
Tactical Data Information Exchange A Subsystem (**TADIXS-A**)/Officer in Tactical Command Information Exchange Subsystem (**OTCIXS**)	These subsystems provide a communications link that exchanges OTH-T information from various shore stations in support of Navy operations.

Table 16-1. UHF SATCOM Major User Subsystem continued

SUBSYSTEM/USER	DESCRIPTION
Demand Assigned Multiple Access Subsystem (**DAMA**)	This subsystem was developed to multiplex several subsystems, or users, on one satellite channel. This has the effect of allowing more satellite circuits to use a UHF satellite channel.
Control Subsystem	This subsystem consists of a communications network that facilitates status reporting and management of system assets.
LEASAT Telemetry, Tracking, and Command Subsystem	This subsystem is a joint operations by Navy and contractors for LEASAT satellite control.
Satellite Monitoring Subsystem	This subsystem provides users of the UHF Satellite Communications System with means to analyze and resolve system equipment related problems. The current subsystem is the Interim FLTSATCOM Spectrum Monitor (IFSM) which will be replaced by the SATCOM Signal Analyzer (SSA).
Tactical Data Information Exchange System Broadcast B (**TADIXS-B**) Tactical Receive Equipment (**TRE**)	This subsystem receives, demodulates, decodes, decrypts, processes and distributes TADIXS B broadcast contact reports.
Tactical and Related Applications Subsystem (**TRAP**)	This subsystem provides near real time contact report data to a variety of TRE users.
Fleet Imagery Support Terminal (**FIST**)	This subsystem transmits imagery from shore locations to ships, ship-to-shore, ship-to-ship, and shore-to-shore.

The major user subsystems listed above are described in greater detail in Section 12, *(Navy Users and Network Architecture*).

16.4 SHF SATCOM/Defense Satellite Communications System (DSCS)

SHF SATCOM provides high capacity, jam resistant, and full-duplex communications for Navy assets both afloat and ashore. It uses communications assets of the Defense Communications System (DCS), in particular the Defense Satellite Communications System (DSCS) SHF satellites. Aside from providing communications services to the Navy community, it provides communications services to the other user communities with divergent requirements. The other communications requirements and users supported include:

- Diplomatic telecommunications
- Airborne command
- Ground mobile forces
- Presidential missions
- CINCPAC secure conferencing
- AUTOVON and AUTODIN trunking
- AUTOSECOM
- NATO

- United Kingdom

Although most shipboard communications are implemented via UHF SATCOM or HF, the inherent nature of SHF and its implementation for the Navy SHF SATCOM System, it nevertheless provides an invaluable communications capability to the Navy. The features provided include:

- High data rates
- Jam resistancy
- Full-duplex communications
- Scintillation free propagation
- Minimum outages due to fading
- Low Probability Of Intercept (LPI) due to narrow uplink transmission beamwidth
- Multi-service and foreign nation interoperability

The DSCS-III system is an integral part of the U.S. Defense communications architecture. It is the successor to the DSCS-II system providing additional satellite channels and exhibiting greater anti-jam capabilities. It provides high capacity SHF-based worldwide communications in support of Command, Control, Communication, and Intelligence (C^3I) information transfer and simultaneously meets both strategic and tactical requirements.

The DSCS system permits small mobile platforms to communicate amongst each other and with command centers via the DSCS satellite constellation. It supports a wide range of user terminals, both mobile and compact, as well as large and fixed installed on airborne, afloat, and earth-based platforms. System terminals range from large earth stations with 20-meter diameter antennas to small airborne platforms utilizing 1-meter diameter antennas.

The DSCS satellites feature antenna coverage patterns which are contoured on command to provide selective terrestrial coverage from shore based control centers. The DSCS III satellites feature a 6-channel communications transponder. Each channel operates with its own RF amplifier. This permits system users to be efficiently grouped from the perspective of the transponder power and frequency spectrum. System users will communicate amongst each other via TDMA and FDMA access schemes.

In spite of these desirable communications features, SHF equipment are relatively expensive due to the complexity of the equipment. Furthermore, their physical size and power consumption far exceed their UHF or HF counterparts. Thus, their SATCOM utility for the Coast Guard is likely to be very low.

16.5 EHF SATCOM

EHF SATCOM System is the latest enhancement to the Navy communications capability. It provides a new survivability capability through its architectural design based on:

- Spacecraft and constellation autonomy
- Nuclear and EMP device protection
- Onboard DSP

The onboard DSP provides Anti-Jam (AJ) and low probability of intercept (LPI) for both the uplink and downlink channels. The Initial Operational Capability (IOC) for the Navy is scheduled to occur in mid-1993. Assets will include two satellites and several sets of equipment for shipboard platforms and shore sites. The satellite's assets will be Fleet EHF Packages (FEP) placed aboard two FLTSAT satellites. The initial EHF capability, the Navy EHF Satellite Program (NESP), will extend EHF communications service to the Over-the-Horizon Targeting (OTH-T) and Navy Tactical Command System -Afloat (NTCS-A) community currently serviced exclusively by the UHF subsystems, OTCIXS and TADIXS. In an evolutionary manner, the EHF capability will eventually supply EHF AJ/LPI service to other tactical users including the SI community. The EHF capability will be implemented under the Copernicus concept via the Communications Support System (CSS) communications architecture. Later in the decade, both the space and earth EHF assets will be expanded with the implementation of the tri-service MILSTAR system. This will provide a full constellation of satellites for worldwide coverage and equipment for both ground forces and aircraft and the communications connectivity between the services.

17 U.S. NAVY USERS AND NETWORK ARCHITECTURE

17 U.S. NAVY USERS AND NETWORK ARCHITECTURE

17.1 Overview

The consumers in today's Navy communications environment exchange data and voice information via dedicated, semi-permanently assigned circuits. The consumers are located on mobile Navy platforms such as ships, submarines, and aircraft, and at ashore communications stations. The stations ashore act as gateways between shore commands and installations (accessed via global DoD land based communications networks such as AUTODIN, Defense Data Network (DDN), and AUTOVON) and the Navy's mobile platforms at sea. Each communications path is associated with a specific data link and network protocol as defined by:

- A fixed frequency assignment
- Specific radio modem and encryption equipment
- Unique message data formats
- A dedicated set of consumers

These consumers are known as "users". The users are Navy personnel or computer systems which have a requirement to exchange text, voice, graphics, and computer -to-computer data. The protocols developed for each of these communications paths were established in accordance with the requirements and funding limitations of each user group. No consideration, however, was given to multiple user groups sharing pooled resources.

17.2 User Community – C^2 Systems

The Navy's C^2 systems disseminate organic and OTH-T battle group tactical data. OTH-T information is communicated via the Tactical Data Information System -A (TADIXS-A) and the Officer-in-Tactical-Command Information Exchange System (OTCIXS) in support of the Navy mission areas of Strike Warfare (STW) and Anti-Surface Warfare (ASuW). TADIXS-A/OTCIXS are mission critical links necessary to employ the Tomahawk cruise missile. Cruise Missile Support Activities (CMSAs), located at USCINCLANT Headquarters in Norfolk and USCINCPAC Headquarters at Camp Smith, HI, plan the missions that are digitally loaded in the guidance unit of the Tomahawk Land Attack Missile (TLAM) and transmit these plans digitally via TADIXS-A to shipboard Tomahawk Weapons Control Systems (TWCS). The locations of enemy ships that could threaten the over water transit of a TLAM or that could be targeted and destroyed with a Tomahawk Anti-Surface Missile (TASM) are transmitted on TADIXS-A by Fleet Ocean Surveillance Information Centers (FOSIC) co-located with CINCUSNAVEUR in London and CINCPACFLT in Makalapa, HI and Fleet Ocean Surveillance Information Facilities (FOSIF) located at Rota, Spain and Kamiseya, Japan. Shore Targeting Terminals (STT) used by Submarine Operating Authorities (SUBOPAUTH) serve as a critical communications guard for re-transmitting OTH-T to

Tomahawk equipped submarines. The STT can receive information via TADIXS-A/OTCIXS and communicate with the submarine either on OTCIXS or on the Submarine Satellite Information Exchange System (SSIXS).

The volume of information transmitted by CMSAs over TADIXS-A can vary significantly. During peacetime, ships and submarines receive the bulk of their TLAM mission data via semi-annual courier delivery of an updated Data Transport Device (DTD). Currently, each DTD can hold approximately 65 Mbytes of mission data. A planned upgrade to the Combat Control System (CCS) MK2 will increase mission data storage to 200 Mbytes aboard Tomahawk capable submarines. Assuming an average of 400 kilobits per mission, this equates to the storage of 1,300 and 4,000 pre-planned missions for ships and submarines, respectively.

NTCS-A also participates on TADIXS/ OTCIXS through their Generic Front-end Communications Processor (GFCP). NTCS-A is used by the Force OTH-T Track Coordinator (FOTC) to ensure a common surveillance picture is available to the Anti-Surface Warfare Commander and each ship in the battle group. In support of the FOTC concept, the Battle Group Database Management (BGDBM) specification outlines three modes of operations for NTCS-A and TWCSs and establishes a dedicated 75 baud FSK broadcast for those ships which are not equipped with a GFCP or a TWCS: controller, participant, and non-participant. Each mode has a different impact on the volume of OTH-T traffic which will be communicated within the battle group via OTCIXS. Normally, the FOTC is the single unit in a battle group operating in the controller mode. In this mode, all ocean surveillance information is received by the FOTC, correlated, and re-transmitted to participant units via OTCIXS and via the 75 baud broadcast. The FOTC correlates information from sensors organic to and remote from the battle group. The FOTC supports the ASuWC in detecting, classifying, and tracking hostile, friendly, and neutral shipping.

17.3 User Community – GENSER Systems

Collectively, the Fleet broadcast (FLTBCST), NAVCOMPARS (source of FLTBCST record data), CUDIXS, and NAVMACS provide the Navy automated General Service (GENSER) Record Message processing. Record Messages are transmitted throughout the shore telecommunications network by using Routing Indicators (RI) to specify the final destinations. Since most afloat commands do not maintain full period telecommunications connectivity, they do not have their own RI. Their traffic is sent to an assigned Broadcast Keying Station (BKS) for fleet broadcast and Record Message delivery. A message delivered to a ship via a direct circuit may also be transmitted on the broadcast. NAVCOMPARS performs a function called "common (broadcast) channel processing" to avoid duplicate deliveries of low precedence messages (priority and routine) to afloat commands. While common broadcast channel processing eliminates many duplicate deliveries, high precedence messages and messages delivered via CUDIXS are exempted from this process, and therefore are prone to multiple duplicate deliveries.

The GENSER portion of the current Fleet broadcast system (the major portion) currently suffers due to the limited throughput of the system. Any one broadcast channel does not exceed a 75 bps data rate. This causes a significant backlog of messages at the BKS NAVCOMPARS, because message data is received from AUTODIN at a 1200 bps rate. A majority of these messages must be delivered on one of these broadcast circuits. The current system design hampers data throughput and operational flexibility on a valuable asset, the SHF Uplink portion of the FLTSATCOM satellite package.

17.4 Major Network Protocols

FLTBCST is the "great holdover" from another era of Navy Communications. With a composite data rate of 1200 bps, with no automated error control, no automated accountability, no flow control mechanisms and its monopoly over the use of the Anti-Jam channels, FLTBCST has been positioned to be the service of last resort. The source of the data has no knowledge of the "integrity" of data received by destinations and, thus, periodically repeats the message information on rerun channels when these channels are available. When Radio Frequency Interference (RFI) is minimal, the Fleet Broadcast provides a reliable means of communications to afloat platforms. When RFI is severe, the Fleet Broadcast becomes backlogged attempting to re-transmit entire messages. Characteristically, as the backlogs and re-transmissions increase, the afloat platforms must process more and more data. There is no flow control mechanism to generate wait or delay transmission states, so more data must be re-transmitted and consequently the delay grows rapidly.

CUDIXS provides a high speed rapid access secure network for the exchange of Record messages between mobile shipboard platforms and the NAVCAMS ashore. CUDIXS operates at 2400 bps with error control and provides shore-to-ship and ship-to-shore connectivity. CUDIXS is implemented in a shore based network controller that interfaces to AUTODIN via the NAVCOMPARS, a network operating over a single satellite channel, and in the NAVMACS afloat. The CUDIXS network protocol has been tailored to achieve a very specific mission – the exchange of Record Messages between a single network control site ashore and many afloat platforms.

The Tactical Intelligence (TACINTEL) subsystem provides essentially the same protocol as CUDIXS with the following exceptions:

- The TACINTEL network has only 24 members instead of 60, as in CUDIXS, and members can directly access each other via the network
- The TACINTEL network accommodates three types of Special Intelligence data (Record Message, BE-3, and OPNOTES)
- The TACINTEL network operates in a Traffic Flow Security (TFS) mode (i.e., no transmission can be correlated to outside world events). Therefore, each TACINTEL transmission is of a fixed size over a given period of time (day, week).

OTCIXS provides rapid interactive data exchange among a large number of network members possessing low throughput requirements. Unfortunately, OTCIXS users tend to use OTCIXS as a "one to many" and "many to one" service, and the corresponding volume of data exceeds that which conforms to the optimal OTCIXS network profile. OTCIXS achieved its ability to accommodate a large population and rapid access at the expense of efficiency. Its design has traded efficiency and high volume throughput in favor of rapid net access by a low volume and large membership population.

TADIXS-A provides a shore-controlled, one-way (shore-to-ship) broadcast of OTH-T tactical ocean surveillance data and tomahawk mission data updates. The broadcast is time-shared among various shore originators. TADIXS-A handles only Tactical Data Processor (TDP) formatted data (OTH-T GOLD). The TADIXS-A shore segment provides for world-wide routing of data amongst shore based facilities. TADIXS-A operates via UHF DAMA at 2400 bits per second.

TADIXS-B is a Navy led multi-service communications system. It is designed to collect and distribute, via UHF SATCOM downlink broadcasts, highly accurate positional and parametric contact data. This information is filtered by receiving equipment and is transferred to designated TDPs for additional processing and display with other OTH-T information. Currently, an interim broadcast called TRAP is being used operationally in place of the TADIXS-B broadcast. Tactical Receive Equipment (TRE) is the terminal equipment which receives the TADIXS-B broadcast. TRE is designated AN/USQ-101(V) and comes in seven configurations. These are:

- (V) 1 – Army
- (V) 2 – Air Force
- (V) 3 – Navy Ship
- (V) 4 – Navy Submarine
- (V) 5 – Marine Corps
- (V) 6 – Navy Shore
- (V) 7 – Air Force (dual channel)

TRE is capable of operating at 2400, 4800, and 9600 bps with or without $\frac{1}{2}$ error coding (symbol rates to 19.2 Kbps).

17.5 Communications Planning Procedures

Procedures governing Navy communications planning and the related topics of frequency management and satellite control are:

- NWP 4: Basic Operational Communications Doctrine. This document describes Navy communications organizations and responsibilities, basic communications planning processes, and the overall methods, media and circuits associated with communications supporting various warfare areas. It also lists and defines all networks available, the frequency bands associated with these networks and the network identifier (line number).
- Fleet Telecommunications Procedures (Instruction C2000.3A). This document describes procedures for the NAVCAMS to assign communications assets, change frequencies and change circuits in response to emergency or catastrophic occurrences such as satellite failure. Candidate frequencies for each service are listed.
- Annex K to CINCxxFLT Operational Order (OPORD) 2000-yr, Fleet Communications Operational Plan. These documents provide Communications-Electronics (C-E) guidance, identify C-E requirements and assign planning and execution responsibilities for communications during joint operations, contingencies, training operations, and day-to-day operations. Interpretation of specific requirements is left to individual commands. Specific Communications tasks, along with special requirements, arrangements and procedures are provided in the communications portions of CINCFLT operational directives.
- Annex K to numbered Fleet command Operational Orders such as COMxxFLT OPORD 201. These documents contain communications plans for various tactical situations or exercises conducted by the numbered fleet commanders. Additionally, they specify the roles of the NAVCAMS; provide COMMPLAN parameters such as required circuits, frequencies, emission characteristics and power requirements; describe how to handle requests for spare line assignments from numbered fleet commands and Battle Force Commands; and describe the procedures for the use of voice circuits.
- Operational Tasking (OPTASK) Communications Messages. These messages contain the detailed communications plan for a specific force or battle group. They indicate frequency assignments for

designated functions/services and guard/monitoring responsibilities. Circuit restoration priorities are also included.

- <u>Ship/Site Communications Plans.</u> These plans are prepared locally for each platform/site and include definition of the specific communications hardware configurations that will be used to implement the directed communications plan.

17.6 Future Directions

The Navy currently has several programs in the formative stages of development which will increase by ten-fold the capacity for automated information exchange. These programs are: SHF/IXS, TD-1389 Emulation, High-Speed Fleet Broadcast, Tactical Intelligence II (TACINTEL-II), MiniDAMA, and the Navy EHF Satellite Program (NESP). These programs will provide additional media, more efficient use of the available bandwidth, additional user access and increased reliability and survivability of the Navy's communications networks. Originally, these systems were designed to provide dedicated independent service to individual user communities without overall system level prioritization.

In order to cohesively structure these new programs, such that the resultant Navy communications system can be attained in an affordable manner and provide the desired survivability and performance attributes, the Navy has elected to develop a new overall command and control architecture. This new architecture has been named Copernicus. Copernicus is both a new C^4I architecture to replace the current system as well as an investment strategy providing a programmatic basis on which to construct the system over the next decade. The Copernicus component which defines the shore-ship-shore communications architecture is the Communications Support System (CSS). The proposed CSS architecture, afloat and ashore, has extraordinary impacts upon several existing and planned programs. It is SPAWAR's intent to implement this architecture through the ongoing development of planned and current systems and that the architecture be implemented with minimal increase to the existing costs and schedules allocated for these individual programs. It is a goal that implementing the Copernicus architecture will improve communications flexibility, responsiveness, throughput, and functionality without any increase in cost. This is made possible by coordinating the efforts of all programs so that products of individual programs are made available to other programs. All new and upgraded Navy C^4I subsystems are planned to be developed using the Copernicus architecture guidelines as a template.

17.6.1 Copernicus Architecture

The Copernicus architecture is comprised of four pillars: the Global Information Exchange System (GLOBIXS), the CINC Command Complex (CCC), the Tactical Data Information Exchange System (TADIXS), and the Tactical Command Center (TCC). As a

C^4I architecture, Copernicus will be constructed as an interactive framework that ties together the command and control process of the Navy tactical commander afloat, the Joint Task Force (JTF) commander, the numbered fleet commander and others with the CINCs ashore.

GLOBIXSs are global, virtual networks imposed on the Defense Communications System (DCS) or commercial systems. GLOBIXSs tie together existing shore sensor nodes, analytic nodes, and other selected activities into communities of like interests. They are by definition joint in construction, and some will be combined.

CCCs are C^4I complexes of multiple Component Command Centers, imposed over Metropolitan Area Networks (MANs) on Oahu, HI, in Norfolk, VA and in Naples, Italy. The CCC will tie together existing command and staff organizations as well as new functional centers such as Space and Electronic Warfare (SEW) center and a Research Center. Viewed from the afloat perspective, the CCC provides a means to manage the information flow for the tactical commander, with sufficient doctrinal and technology flexibility to allow each commander to decide how much and what kind of information he wants. Thus, the afloat commander should see the CCC as a group of shore-based assistants somewhat analogous to the Composite Warfare Commander (CWC) of the Carrier Battle Group (CVBG) afloat. In the same way the CWC commanders are delegated war fighting ends and means afloat, the CCC will have analogous personnel to whom ends and means ashore may be delegated.

TADIXSs are a series of virtual nets that support afloat-ashore connectivity. Copernican TADIXSs are not to be confused with the existing TADIXS A and B. Rather, Copernican TADIXSs are virtual networks of variable duration depending on the information exchange load. TADIXS should not be considered communications circuits, but information networks sharing communications circuitry over a broad menu of bearer services from HF and VHF to UHF, SHF, and EHF military satellites, as well as commercial satellites.

The final pillar of the architecture is the TCC, which is intended to be a generic term reflecting the nerve centers of tactical forces/units – whether carriers, submarines, or Marine Air/Ground Task Forces (MAGTF) in the Navy-Marine model, or Corps, Air Wings and JTFs in the joint model.

The key feature of the Copernicus architecture is that the operations user is at the hub of the information universe. In today's information scenario, the operations user has information "pushed" to him by the shore establishment whether he needs it or not. With the advent of Copernicus, the operational user will be able to "pull" the information he needs by specifying these needs in advance.

17.6.2 Communications Support System

Operationally, the CSS communications architecture will enhance battle force communications connectivity, flexibility, and survivability through multimedia access and media sharing. CSS permits users to share total network capacity on a priority demand basis in accordance with the current communications plan. Automated network monitoring and management capabilities are also provided by CSS to assist operators in the real-time allocations of communications resources. Some advantages to this approach include significantly increased communications survivability without sacrificing user throughput or communications efficiency. This is achieved by using the automated access to multiply media and very fast switching times possible with CSS. Communications over another RF medium which is jammed can be automatically rerouted over another RF medium, normally so quickly that the user communications are not disrupted at all. Existing communications equipment can be better used through load sharing. New communications capabilities can be incorporated without requiring expensive changes to the user's baseband equipment or operating procedures. In addition to reducing expense for equipment upgrades, this will allow existing systems to use improved communications equipment without the prolonged time lag currently required for modifying user baseband equipment. New users can be accommodated on a priority basis rather than an absolute (have or have not)basis. No new radios are required to support a new user unless the total capacity required for a site is exceeded.

CSS resources consists of the media (channel, frequency, time slot, etc.), the communications equipment (radios, modems, link encryption units, multiplexers, etc.) employed to effect communications on that medium, and the communications protocols which provide access, routing and control of the exchange of information between the nodes operating on that resource. The addition, deletion, or modification of communications resources on an asset require only hardware interface modifications and system communications plan updates. Each user can share multiple resources. New protocols will permit users to approach the communications system as a pool of resources available to effect data transfer to and from other platforms. The overall communications system will, as a result of this architecture and the embedded protocols, be capable of providing more timely and more accurate data transfer across a broad range of operational environments.

Perhaps best of all, the required changes can be accomplished with minimum impact upon funding profiles for existing and planned programs. The CSS modularity will substantially reduce development and life cycle support costs of future systems (such as TACINTEL II, TACINTEL II+) by the use of open system architecture principles.

As with today's communications, the communications plan is central to the functioning of CSS. The CSS communications plan is similar to current communications plans. CSS will provide automated tools to help communications planners adapt standard COMMPLANs to CSS COMMPLANs.

Where possible, the system architecture had been carefully designed to be compatible with existing organizational procedures. The steps involved in communications planning and system operations are nearly identical to those currently performed with existing equipment. CSS will require few, if any, new personnel, and basic job functions of existing personnel will not significantly change. The most significant change will revolve around the increased ability of command personnel to adapt the site's communications capability to a changing operational environment. In short, the CSS demonstrates that modern technology can be applied to reduce costs and improve capabilities in Navy communications.

18 INTEROPERABILITY ARCHITECTURE

18.1 Overview

The Next Generation Cutter (NGC) communications architecture will rely significantly on the use of COTS and Non-Development Items (NDIs). This approach will permit present and future Coast Guard communications requirements to be met at a substantial cost savings. Cost savings will be realized by a combination of reducing operating costs (power consumption), standardizing software (operating systems, graphical user interfaces, communications protocols), and lowering hardware/software costs (competition in the commercial market).

An "open" architecture system will be the backbone for the NGC communications architecture. An open system is one which can adapt to divergent operational requirements, both now and in the future. An open system enables new technology and applications to be seamlessly integrated. An open system can best be characterized by:

- Rapid implementation
- Flexibility
- Adaptability
- Low life-cycle maintenance costs

This section presents draft recommendations resulting from a survey of existing communications system technology and the anticipated overall communications requirements for the next generation of Coast Guard cutters. This Section looked at requirements from the point-of-view of a generic afloat platform; it provides high-level technical recommendations for the requirements to be incorporated in the procurement of the next generation of Coast Guard cutters.

18.2 Intraship Communications Architecture

The NGC will implement an integrated distributed environment in support of the cutter's communications mission. This distributed environment will be one in which a heterogeneous collection of individual, autonomously executing, processing elements are integrated by a Standard Operating Environment (SOE). These processing nodes will range from integrated workstations to high-performance single-board computers.

The physically disparate computing elements within the NGC intraship communications framework will be tied together via a FDDI ring. The main functional components implementing the NGC's communications requirements will be:

- External link communications
- End-to-end communications
- System control and management
- Support services

Figure 18-1 illustrates the component architecture of the SOE-based NGC intraship communications architecture.

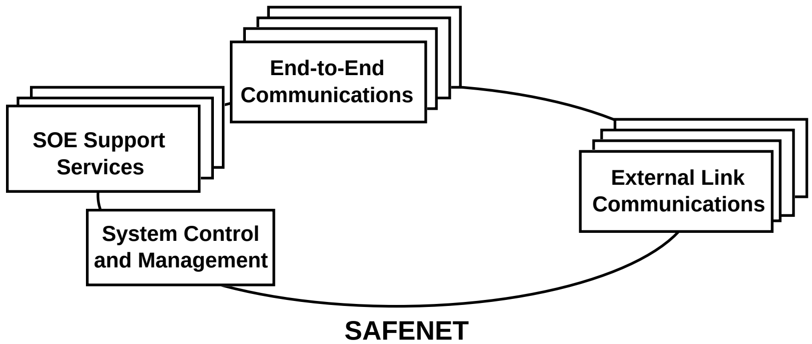


Figure 18-1. NGC Intraship Communications – Component Architecture

18.2.1 NGC Standard Operating Environment

The SOE integrates the different communications applications working on individual parts of the overall communications mission. The SOE provides the interface mechanisms to these processing nodes (full-featured workstations and high-performance, real-time single-board computers, as mentioned earlier) such that applications operating on these individual processing platforms can communicate amongst themselves and can obtain system-wide services. It supports the synchronization and communications requirements of these peer communications applications (client to client) and provides access to system services and resources (client to server). The SOE also defines interfacing conventions such that small personal computers can also be configured to use the services provided by the SOE. Figure 18-2 illustrates the overall generic architecture of the NGC's intraship communications architecture.

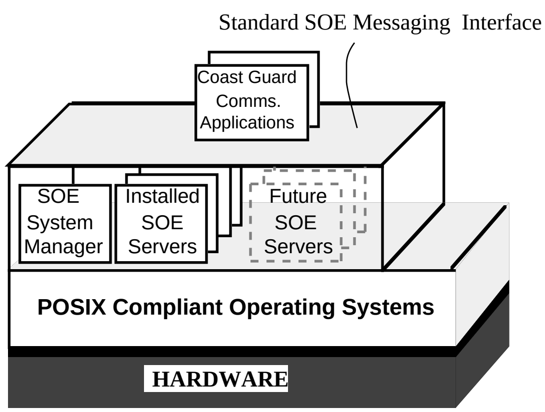


Figure 18-2. NGC Intraship Communications – Functional Architecture

The processing nodes within the intraship communications system will be interconnected through the interprocess communications capability of the NGC SOE. The SOE will implement this capability based upon the Survivable Adaptable Fiber Optic Embedded Network (SAFENET-II) protocol profile. Both the standard OSI TP4/CLNP-based stack, as well as the Express Transport Protocol (XTP)-based light-weight protocol stack, will be implemented. The use of the XTP stack will be restricted to the interprocess communications requirements of nodes involved in time-critical equipment control operations. The following subparagraphs describe each component of the NGC SOE.

18.2.1.1 POSIX Compliant Operating System

The underlying collection of operating systems controlling the standard operating environment will be POSIX IEEE 1003 compliant. However, their POSIX features will be implemented in three separate profiles as determined by their "real-time" responsiveness requirements. At one extreme, the full-feature workstations will be fully POSIX compliant as specified by IEEE 1003.1. The other two profiles will be a subset of the real-time version of POSIX as specified by IEEE P1003.4. These two subsets will be the "Real-Time Controller System" and the "Multipurpose Real-Time System" profiles as defined by the POSIX document IEEE P1003.13.

18.2.1.2 SOE Common Operating Interface

An inherent property of distributed systems is the separation of its components. The SOE will address the consequences of component separation (i.e., the explicit need for system communications, management, and integration capabilities) with a system design which will provide for the following forms of *transparency*:

- <u>Access transparency</u>. Local and remote objects (e.g., files, printers) are accessed by identical operations.
- <u>Concurrency transparency</u>. Multiple applications operate upon shared data objects without interference and without causing inconsistency in the data.
- <u>Location transparency</u>. Objects are accessed without any knowledge by application processes of their physical location.
- <u>Migration transparency</u>. Objects may be moved within the SOE without affecting the operation of applications.
- <u>Performance transparency</u>. The SOE, as directed by an operator, may be re-configured to improve or maintain performance based upon the processing load.

This transparency will be implemented through a "messaging" interface whereby the underlying physical characteristics of the system are hidden to the various Coast Guard communications applications. This "messaging" interface will provide a generic interface to system entities (e.g., other applications, the database service). This will provide a key interoperability feature to the NGC's communications system. For instance, as applications are developed to use databases according to the conventions defined by this "messaging" system, these same applications will be able to execute without modification should the Coast Guard implement an upgraded database from a new vendor in the future. The SOE "messaging" interface will provide Structured Query Language (SQL), Graphical User Interface (GUI), and Interprocess Communications (IPC) independence.

18.2.1.3 Human Interface Control Service

The SOE will provide a user-friendly interface which will manage operator interaction with any number of communications applications. The interface environment will be composed of the following three elements:

- An input selection model
- A window manager
- Application programs

This environment will be based upon an object-action input selection model. The selection model defines the actions that users must perform to control the window manager and applications in this environment. The selection model follows a point-and-click paradigm. Users first point at and select an object with which to work, and then point at and select an action to perform on the selected object.

The window manager will provide users with a way to manipulate the windows displayed in the interface environment. Typical manipulations include opening, moving, closing, switching to, sizing, and minimizing/maximizing windows and arranging them as required on the display.

The window manager will frame application windows with an eight-segment border that can be stretched to resize the window. A title area supplied by the window manager will display a title for the window and can be used to move the window. Graphical buttons embedded in the window manager frame will provide a window management menu and other window controls. The window manager will have a three-dimensional appearance so that the control buttons, when "pressed" by the pointing device pointer, actually look like they have been pressed. The window manager will provide for consistent behavior from one application to the next.

Application programs will fill the space inside the window frame. All applications will follow the guidelines specified by the latest version of the standard GOTS Style Guide. The application's behavior will be consistent with the behavior of all other applications in the standard Coast Guard operating environment. Figure 18-3 illustrates a sample base window for the SOE.

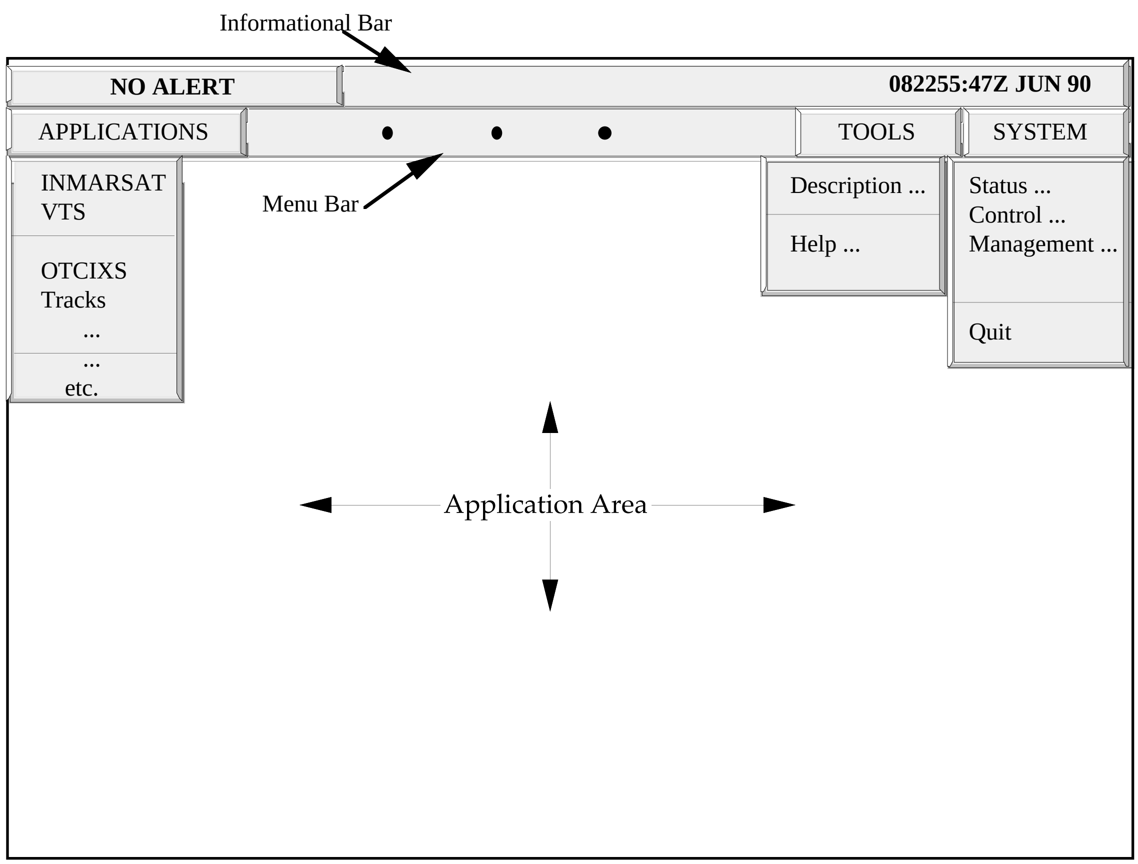


Figure 18-3. Sample SOE Base Window

The *Applications* menu will list the names of the Coast Guard communications applications available within a system. Selecting an option from this menu will activate the application and open its primary window. Figure 18-4 illustrates the standard components of a primary window.

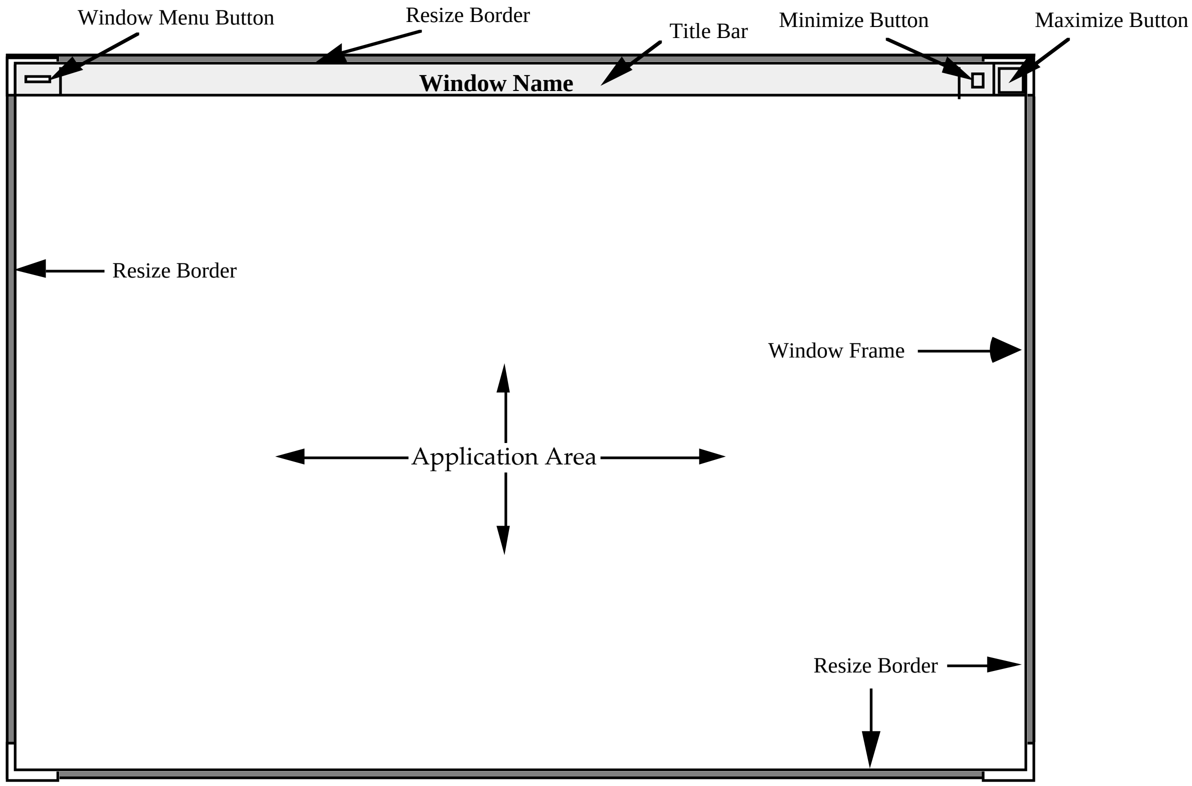


Figure 18-4. Coast Guard Application Primary Window

A graphic print capability will be provided to applications by the SOE whereby the contents of any window can be sent by an operator command to a system print device. An "alerts" capability will be provided by the SOE which controls the specification of events or deficiencies about which the user wants to be informed and provides access to alerts that have been received.

18.2.1.4 Database Management

This is an SOE service which will be available to any Coast Guard application requiring its services. It will be implemented with a COTS relational database which meets the following requirements:

- Structured Query Language ISO 9075 "Database Language SQL With Integrity Enhancement"
- 4GL support
- Data integrity implementation and enforcement within the Database Management System (DBMS)
- Capability for data-triggered events within the DBMS
- X Windows compatible user interface development tool allowing transportability of applications across UNIX/POSIX environments

- Capability to provide database services to remote clients over the network via standard Operating System (OS) capabilities (e.g., remote procedure calls)
- MOTIF™ style GUI tool with the ability to display graphics pictures

18.2.1.5 Expert Inference Service

The SOE will provide an expert system server to provide an inference service to communications applications which may require such a service. "Real-time" applications using this service will send to this server a variable stream of raw information for inference processing. This service will subsequently return a response based upon the set of "rules" earlier provided by the application to the server. Sample applications served by the use of the Expert Inference Service will include:

- Anomaly detection/resolution
- Fault diagnostics
- Fault avoidance
- Intelligent control of Radio Frequency (RF) equipment

18.2.2 Communications Components

The intraship communications architecture of current cutters is not well integrated and lacks the reliability, control, and flexibility required to provide multipath connectivity for current and future communications traffic requirements. Consequently, the NGC will implement a modular and integrated architecture which provides radio room and Combat Information Center (CIC) operators with enhanced communications management capability. The NGC intraship communications architecture will be based upon a hierarchically distributed architecture (local area network, back plane bus) which will provide flexible control of shipboard communications assets, real-time display of communications system status, and improved overall reliability.

The NGC's communications applications will be based upon an open system architecture utilizing VMEbus technology. Versa Module Europe (VME) is the current industry standard for high-performance computing requirements. This technology will allow the Coast Guard to maximize use of COTS hardware technology. VME allows flexible component configuration and interconnection with both VMEbus-based components and non-VME components via local area networks. This is an architecture which permits multiple vendors to provide boards, cards or module hardware and software and ensures operational compatibility between these COTS components.

The VME modules will be used for all external communications/processing functions of the NGC and will process Coast Guard signals in analog, digital, and RF formats. The communications applications include its use for LOS, Beyond Line-of-Sight (BLOS), and SATCOM channels. When the VME Chassis is combined with UHF, HF, VHF, EHF, or SHF receiver and exciter VME-based functions, modem-based functions and various interface/control boards, it will form the communications processing element for the NGC's external communications system. These multimodule units will form a flexible and adaptable receive and processing system.

The VME modules will be contained within a standard VME Chassis. The standard Coast Guard VME Chassis will be a 19-inch-based rack-mountable unit as specified by EIA-RS310-C-77. The VMEbus modules it can accommodate will be in the 9U (220 mm) and 6U (160 mm) VME form factors. It will accommodate standard VME modules having 32-bit data and address bus (or less non-multiplexed or 64 bits multiplexed, i.e., VME64). The chassis housing will contain all emissions and spurious signals to prevent interfering radiation from emanating into or out of the enclosed modules. Two main functional types of VME modules will be supported in the NGC computing architecture. These are:

- Digital Module. contains digital processing components, memory or digital interface devices as follows:
  - *Processor*. Performs generic data or iterative processing
  - *Memory*. Used to store data or information
  - *Interface*. Used to interface with external equipment
- Analog Module. contains analog processing components such as filters, operational amplifiers, IF amplifier, and frequency synthesizers as follows:
  - *RF*. Processes analog signals above 80 MHz
  - *IF*. Contains analog components (maximum frequency of 80 MHz)
  - *Baseband*. Processes analog signals (maximum frequency of 1 MHz)

Table 18-1 lists the types of VMEbus modules to be used in the NGC.

The chassis will be capable of supporting ELF/LF/MF/HF/VHF/UHF/SHF exciter and receiver operations. Furthermore, with the addition of the RF analog modules and the use of an external power amplifier, it will provide a similar RF interface for half-duplex or full-duplex transceiver operation. The I/F processing and chassis control/interface modules will be capable of providing baseband interfaces to support all required operating modes, such as SATCOM, LOS, and BLOS modes. The chassis will provide the interface connection capabilities, the VME power, the bus interface, and the mechanical interfaces to support the VME baseband interfacing modules. A modular VMEbus-based architecture would typically be configured on board the

NGC to implement the external link communications capability of the NGC (as illustrated in Figure 18-5).

Table 18-1. VME Module Categories

TYPE	FUNCTION	OPERATING CHARACTERISTICS
Digital/Processor	General processing	Processing for equipment control and communications protocols to effect external communications (ISO layers 3,4)
Digital/Processor	DSP	Incorporates digital signal processing as the primary basic function
Digital/Processor	MUX	Processor used to multiplex or demultiplex
Digital/Processor	Control	Used for communications controller
Digital/Memory	Dynamic	Storage module, volatile
Digital/Memory	Static-non-volatile storage	Storage module with nonvolatile memory such as EEPROM, EPROM, or battery backup SRAM
Digital/Interface	Serial	Module having serial interface and functions as an interface element (up to T1 rates, electro or optical I/F)
Digital/Interface	Ethernet	(IEEE STD 802.3/ISO 8802-3) interface
Digital/Interface	SAFENET II	MIL-HDBK-0036 compatible interface board
Digital/Interface	Intelligent I/O	Module having multiple serial or parallel interfaces each being reconfigurable and performing the first and second ISO layers
Analog/RF	Rcvr/Excit	Contains both receiver and exciter
Analog/RF	Receiver	Receiver module, output is either an IF or baseband/disk and/or combiner
Analog/RF	Exciter	Exciter module, output is either an IF or baseband/disk and/or combiner
Analog/IF	Modem	Modulator and demodulator
Analog/IF	Demod	Demodulator, output is baseband
Analog/IF	Signal Combiner	Maximal ratio prediction or post-detection
Analog/Baseband	Interface	Analog interface – either input or output

The communications applications executing on the VME modules are segregated into two groups: those performing low-level functions (e.g., equipment control) and those performing high-level communications processing (e.g., external communications protocols). In keeping with the modular and open NGC architecture, a standard interface protocol will be defined for communications between these disparate components. This protocol will specify the elemental data elements which will be passed to/from the equipment control components. Examples of information transmitted include items such as operating mode, modulation, frequency, and control parameters. Figure 18-6 illustrates the standard interface according to the OSI layered abstraction.

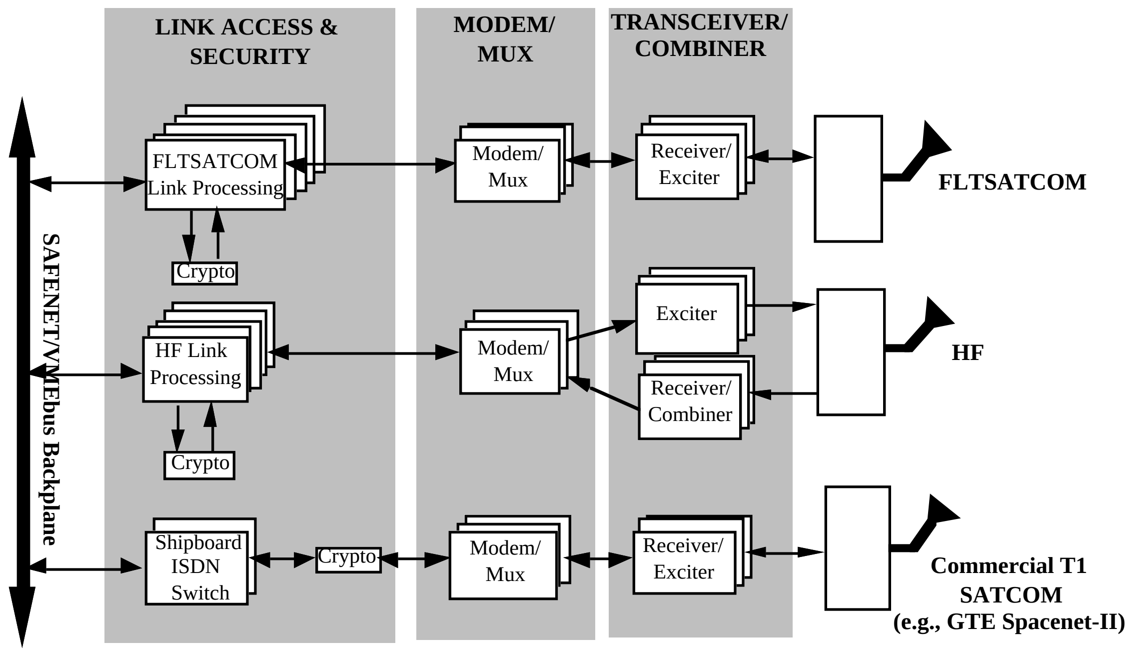


Figure 18-5. Next Generation Cutter – Sample of External Link Communications Architecture

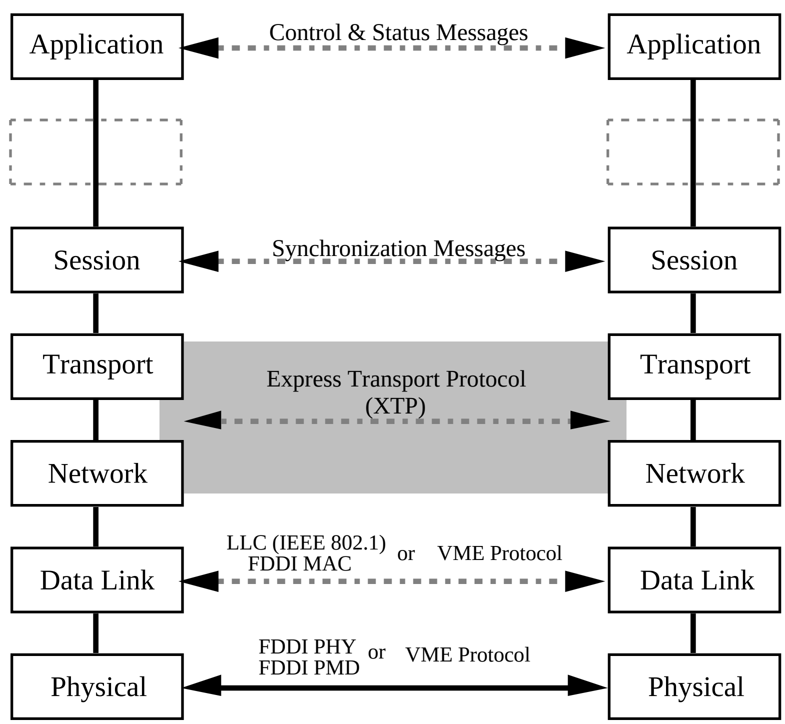


Figure 18-6. Protocol Architecture – To/From Equipment Control Component

18.3 External Communications Architecture

The communicating entities within the external communications architecture for the NGC will function in an organized and cohesive environment. It will incorporate existing, evolving, and planned communications subsystems and devices. It will use both commercial and U.S. Navy communications architectures such as the Navy's CSS and GTE's SPACENET-II commercial domestic satellite. It will incorporate the latest network technology such as multicast/lightweight transport protocols, distributed databases, and transmission resource sharing into its architecture. Figure 18-7 depicts the communications architecture for the NGC.

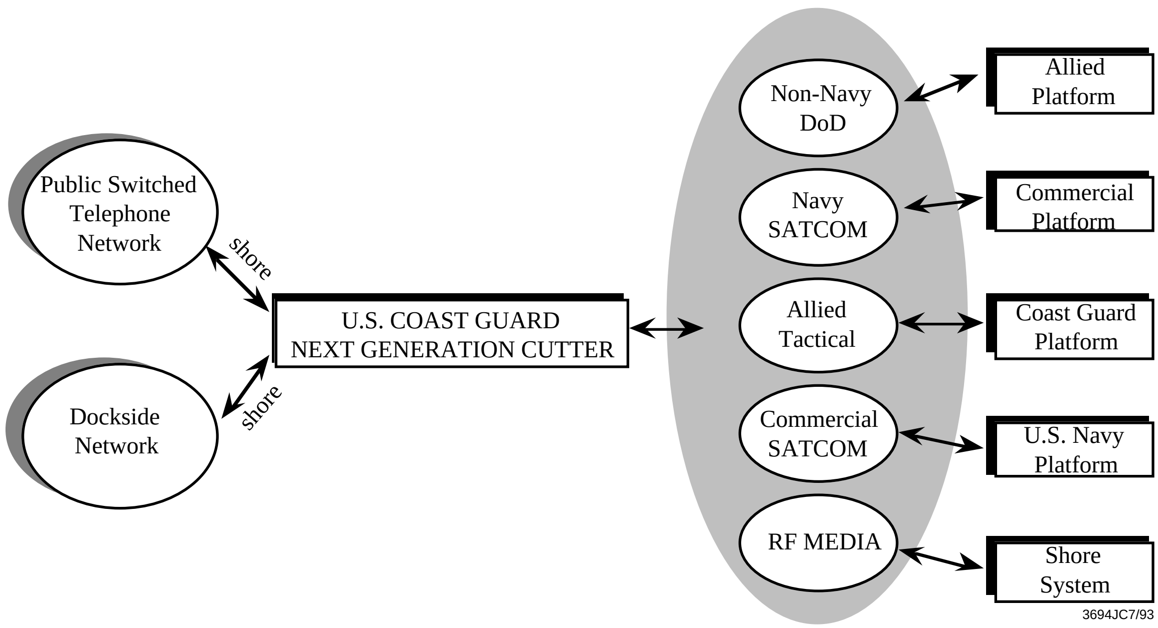


Figure 18-7. U.S. Coast Guard Next Generation Cutter – External Communications Architecture

18.3.1 Communications Protocols

The overall intersite communications network (i.e., between shore, aircraft, and afloat platforms), within which the NGC will communicate, will consist of a widely disparate collection of subnetwork technologies. These technologies range from commercial ISDN subnetworks and commercial satellite-based T1 links to existing Navy and Coast Guard RF resource assets (i.e., UHF SATCOM, SHF SATCOM, HF LOS/ELOS, etc.). The external communications network must support all Coast Guard operations. This operational environment is characterized by information exchanges in support of many different missions. Voice, tactical data, record messages, imagery, and E-mail data must all be transmitted over this network. The scope of these operations imposes a requirement for a wide range of communications transport services on the external communications network.

In general, the external communications network architecture for the NGC must support the following types of information exchanges:

- Support and Planning. Support and planning information exchange is effected by messages ranging from traditional naval Message Text Format (MTF) and Bit-Oriented Messages (BOMs) to imagery data. These messages may be either tactical or non-tactical. Typical users will include planning and decision-making systems such as the Joint Operational Tactical System (JOTS), as well as those systems involved with the exchange of administrative information. The required delivery times for these messages are on the order of hours and days.
- Command and Control. Non time-critical command and control information exchange is effected primarily by COMs. However, special message formats such as OTH-T Gold and Fleet Numerical Weather Center MTF are also used. The required delivery times for these messages are on the order of tens of minutes and hours.
- Tactical Coordination. Tactical coordination messages primarily contain computer-to-computer data but also include communications based intelligence summaries and BOM Links. Ultimately, these are processed by tactical information systems for display to an operator. Tactical coordination messages also include Operator Notes (OPNOTES) exchanged between TDP operators. The required delivery times for Tactical Coordination messages range from seconds to hours.

These heterogeneous subnetworks will be tied together by a common glue. The interconnection glue will be the OSI CLNP as specified by ISO 8473. The NGC external communications protocol architecture will conform to the OSI communications protocol standards. The use of existing off-the-shelf OSI protocols for this end-to-end transfer of information will be maximized. However, some new protocols will need to be developed because none of the currently available COTS OSI protocols meet certain Coast Guard throughput and delivery requirements, due either to the excessive overhead generated by these protocols or the lack of some necessary functionality. A primary area of concern with respect to the existing OSI protocols is tactical data which must be rapidly transmitted and received and, possibly, rapidly responded to. In particular, existing OSI protocols are functionally deficient to meet the needs of tactical information exchange due to the following characteristics:

- Lack of multicasting capabilities

- Packet formats and "header" field layouts tailored for elegant software manipulation at the expense of great overhead
- Lack of "accountability" mechanisms
- Architectural design based implicitly on "cheap" high bandwidth networks, as opposed to the characteristically low and "expensive" data rates of current RF networks

The NGC external communications architecture will adhere to the ISO 7-layer model. This architecture will be primarily geared to support the use of existing off-the-shelf OSI protocols. Any protocols specially developed to meet the requirements not met by existing COTS OSI technology will themselves conform to OSI standards and will be interoperable with the off-the-shelf OSI protocols.

The core functional elements of the NGC external protocol architecture will implement the functionality of the network and transport layers of the OSI model. The network layer will be implemented with the following protocols:

- ISO CLNP. This is the CLNP as defined by ISO 8473.
- Modified CLNP. This is a network protocol based upon the ISO 8473 which will provide a superset of ISO 8473 functionality. It will add multicast addresses and provides node-to-node delivery accountability. This protocol will be a developmental item.
- ISO ES-IS. This is an End System-to-Intermediate System (ES-IS) protocol specified by ISO 9542. It will be implemented to provide up-to-date information for the ISO 8473 CLNP routing algorithms.
- ISO IS-IS. This is an Intermediate System-to-Intermediate System (IS-IS) protocol specified by ISO DIS 10589. It will be implemented at sites providing "gateway" functions. These sites will primarily be Coast Guard shore centers. However, the NGC may also perform gateway functions for aircraft-to-shore communications. This protocol will be used to provide other gateways with raw information about every other gateway in the network and the state of its directly connected (subnetwork) "links".

The ISO 8648 network architecture functionally decomposes the network layer into three components. These are:

- Subnetwork Independent Convergence Protocol (SNICP)
- Subnetwork Dependent Convergence Protocol (SNDCP)
- Subnetwork Access Protocol (SNAcP)

The SNICP implements internetwork routing and relaying and other functions necessary to transfer data between subnetworks. The SNICP will be implemented in the

NGC communications architecture by the ISO 8473 CLNP. The SNAcP is a media-dependent component. It implements the routing and relaying of data within a subnetwork. The SNDCP is an interface component and will provide a common interface between the "standard" ISO 8473 CLNP and the media specific SNAcP. Figure 18-8 illustrates the ISO 8648 network architecture which will be used for the external communications network.

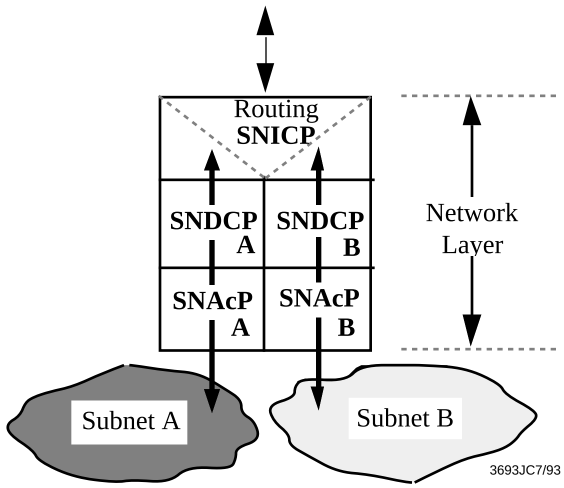


Figure 18-8. ISO 8648 Network Layer Architecture

The transport layer will be implemented with ISO Transport Protocol 4 (TP4), as specified by ISO 8073, and with a new lightweight connectionless transport protocol based upon the ISO Connectionless Transport Protocol (CLTP), as specified by ISO 8073 Addendum 2. This enhanced CLTP will be a developmental item. The application layer will contain an Accountability Management process which will be responsible for obtaining and providing end-to-end message accountability information to those Coast Guard services requiring this information. Figure 18-9 depicts the network protocol architecture for the NGC external communications network.

The modified CLTP-based transfer service (stack) will be a scaleable transport protocol in that the communicating "users" of the transfer service are able to configure the transport functionality. They will be able to trade off performance against speed. At the low end of the performance curve, the transfer service will not provide any reliability (e.g., error control, sequencing). However, it will provide accountability. Accountability refers to the fact that the transmitter of a message will always be given at least a tacit indication that a particular message was transferred, with a high degree of confidence, to its destination. It does not, however, imply that the message was received error free. This modified CLTP-based stack will be used when speed is of the essence. It will be designed to work equally well over both high-speed and low-speed RF paths.

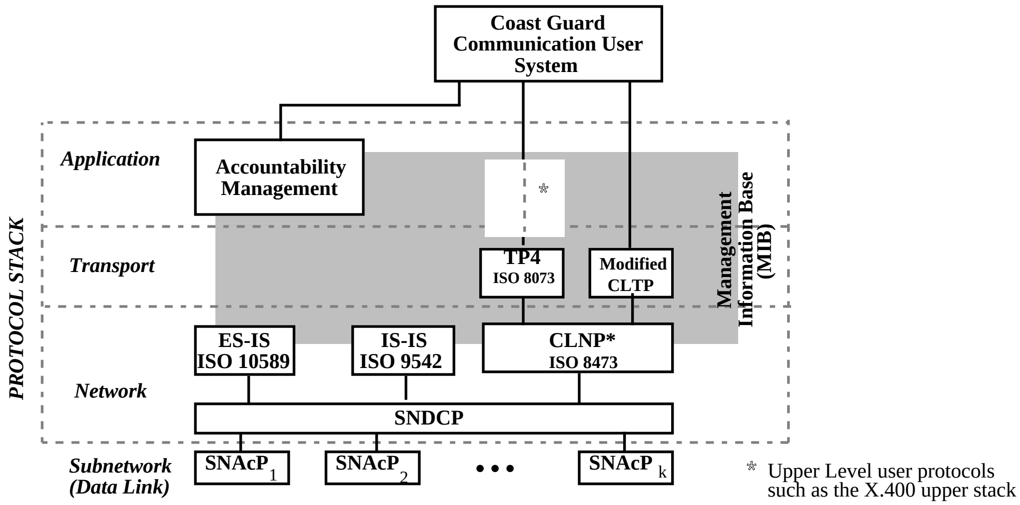


Figure 18-9. Overall Network Protocol Architecture

The modified CLTP will be capable of implementing some or all of the following end-to-end delivery functions:

- Error Detection and Correction. Messages received by the destination transport service user(s) are free from bit errors.
- Sequenced Data Delivery. Messages received by the destination transport service user(s) are sequenced as presented by the originating transport server user. Any duplicate data is discarded.
- Complete Data Delivery. Message data presented to the transfer service is transferred completely to the destination transport service user(s) without any data "gaps".

The Error Detection and Correction function, by itself, does not ensure that a received message is free from duplicate information or that its contents is ordered as presented by the originating transport service user. The Sequenced Data Delivery function, by itself, does not ensure that received messages are free from bit errors and are not missing any data fragments. The Complete Data Delivery function, by itself, does not ensure that received data is correctly sequenced, is error free, and is free of duplicate data. The following classes of transport services will be provided by the modified CLTP to the transport "user":

- Basic – Class 0. This transport service provides NO Error Detection and Correction and NO Complete Data Delivery but provides for Sequenced Data Delivery. This will be the least reliable transport service provided by the modified CLTP. Class 0 will be best suited for transfer service "users": 1) who exchange very "small" messages ("small" in the sense that the transport protocol is not required to perform segmentation and reassembly functions); 2) who have no internetting requirements; and 3) for whom rapid delivery takes much higher precedence over Error Detection and Correction. This basic transfer service may be used to transfer low-quality video images for which gaps and errors may be perfectly acceptable. No error detection and recovery and, thus, no retransmission protocol mechanisms are used by this protocol class. This protocol class will impose minimal transport layer overhead.

- Error Free – Class 1. This transport service provides both Error Detection and Correction and Sequenced Data Delivery. This is suited for services for which the delivery of error-free and correctly sequenced messages with "gaps" is acceptable. An example may be text messages in which the content of the message may be understood in spite of missing "gaps" of information.
- Basic Reliability – Class 2. This transport service provides NO Error Detection and Correction but provides for Sequenced Data Delivery and Complete Data Delivery. This would be suited for transfer service users who exchange "large" messages ("large" in the sense that the transport protocol is required to perform segmentation and reassembly functions) and for whom complete messages are required though bit errors in the message are tolerated. An example application for this service may be moderate quality video information in which moderate levels of errors (i.e., noise) are tolerable.
- High Reliability – Class 3. This transport service provides full reliability (Error Detection and Correction, Sequenced Data Delivery, and Complete Data Delivery). This class is suited only to those transport service "users" willing to accept the performance penalty imposed by the protocol's reliability mechanisms.

18.3.2 RF Equipment

The following constraints will be imposed upon the communications infrastructure for the NGC:

- Minimizing equipment weight is paramount.
- Equipment must be capable of withstanding shocks of up to 5 gravitational units (Gs).
- Operational temperatures will range from 10° to 45°C.
- Primary power provided by the Cutter is 115 or 230 volts VAC at 60 Hz; an outline backup power supply will be switched no later than 0.5 seconds after primary power failure.
- The Cutter's radio room must provide sufficient physical space and appropriate layout so as not to impose any constraints upon the arrangement of existing radio equipment technology.

Based upon these constraints, and keeping with the VMEbus-based implementation of the internal NGC communications architecture, the physical equipment controlling the external subsystem of the NGC communications architecture will be based upon a modular open architecture. Furthermore, it will take advantage of the substantial technological advances which have occurred recently; these advances are characterized by enhanced performance and reliability over the previous generation of similar equipment and subsystems. In addition to performance and reliability, the equipment will feature power and physical space requirements far less than that associated with the technology of current cutters.

18.3.2.1 SATCOM Antenna

Substantial topside space and weight is required to install SATCOM antenna systems on board U.S. Navy vessels. Because the NGC must have even greater interoperability capabilities with U.S. Navy systems than is presently the case, the NGC will employ a multiband SATCOM antenna in order to minimize topside space and weight. These antenna systems will combine the operations of multiple frequency bands into a single antenna system.

18.3.2.2 HF Communications

Present-day conventional HF systems deployed onboard afloat platforms for long-range communications require high-power transmitters and larger, bulky receivers. These components also impose an extensive power requirement. Furthermore, the antenna subsystems require substantial physical space. The NGC HF ELOS communications subsystem will employ compact RF equipment and will automate much of the manually intensive requirements imposed upon operators today to maintain HF circuits.

18.3.2.2.1 HF Receiving Subsystem

The HF receiving subsystem will provide a high gain and omnidirectional HF receive capability. This will be implemented with multiple receive antennas and automated adaptive processing capabilities. The NGC will use state-of-the-art DSPs to mitigate the effects of atmospheric noise, multipath effects, and antenna pattern nulls to enhance the S/N (signal-to-noise ratio) of HF signals. This subsystem will selectively search for a predefined signaling pattern and filter out any unwanted signals. The signals from the multiple antennas will be combined and processed such that the antennas are "steered" toward the intended signal emitter, thereby increasing the signal gain.

18.3.2.2.2 HF Transmitting Subsystem

The HF transmitting subsystem will provide a fully automated transmit beam steering capability. Based upon inputs from the direction-finding capabilities of the

NGC's HF receiving subsystem, this subsystem will automatically determine the appropriate signal transmission angle for optimal ionospheric diffraction to the destination site. The transmit beam will be bidirectionally steerable (azimuth, elevation).

18.3.2.2.3 HF Link Establishment

In order to mitigate the effects of atmospheric interference and "dead zones", the HF transmitting subsystem will provide Automatic Link Establishment (ALE). It will be capable of automatically selecting an optimal frequency from a set of frequencies for transmission to a destination site.

18.3.2.2.4 HF Modem

The NGC will implement a high-performance single-tone modem. The associated wave form will adhere to the single-tone 2400 symbols per second phase modulated (8-ary PSK) wave form specified by MIL-STD-188-110A. This modem will be capable of adaptively removing most multipath effects and will possess the capability to perform convolutional error correction.

18.3.2.3 Dockside Information Exchange Interface

The NGC will possess an interface capability to exchange information with ashore Coast Guard facilities. The data will be exchanged over unsecured public networks including LANs, MANs and WANs. End-to-end encryption of data will be performed over these public networks, and the interface will adhere to commercial ISDN Basic and Primary rates and signaling formats.

APPENDIX A GLOSSARY OF TERMS

AC	Alternating Current
A/D	Analog to Digital
ADM	Adaptive Delta Modulation
ADPCM	Adaptive DPCM
AFSK	Audio Frequency Shift Keying
AJ	Anti-Jam
ALGaAs	Aluminum Gallium Arsenide
AM	Amplitude Modulation
AMVER	Automated Mutual-Assistance Vessel Emergency Rescue
AMTOR	Amateur Teleprinting Over Radio
ANSI	American National Standards Institute
ARQ	Automatic Repeat Request
ASK	Amplitude Shift Keying
ASuW	Anti-Surface Warfare
AT&T	American Telegraph and Telephone
ATM	Asynchronous Transfer Mode
AUTODIN	Automatic Digital Network
AUTOSEVOCOM	Automatic Secure Voice Communications
AUTOVON	Automatic Telephone
BER	Bit Error Rate
BGDBM	Battle Group Database Management
BISDN	Broad Band ISDN
BKS	Broadcast Keying Station
BLOS	Beyond Line-of-Sight
BOM	Bit Oriented Message
bps	bits per second
BPSK	Binary Phase Shift Keying
C^3I	Command, Control, Communications and Intelligence
CCC	CINC Command Complex
CCEP	Commercial COMSEC Endorsement Program
CCITT	Comité Consultatif International de Télégraphique et Téléphonique
CCS	Combat Control System
CDMA	Code Division Multiple Access
CES	Coast Earth Station
CFDM	Constant Factor Delta Modulation
CGDN	Coast Guard Data Network
CGSWS	Coast Guard Standard Workstation
CIC	Combat Information Center

CINCLANT	Commander-in-Chief, Atlantic
CINCPAC	Commander-in-Chief, Pacific
CINCUS NAVEUR	Commander-in-Chief, U.S. Naval Forces, Europe
CLNP	Connectionless Network Protocol
CLTP	Connectionless Transport Protocol
CMOS	Complementary Metal Oxide Semiconductor
CMSA	Cruise Missile Support Activity
COMMPUSEC	Computer Security
COMMSTA	Communications Station
COMSEC	Communications Security
COMSAT	Communication Satellite
CONS	Connection-Oriented Network Service
COTS	Commercial-Off-the-Shelf
CPFSK	Continuous Phase FSK
CPU	Central Processing Unit
CSMA	Carrier Sensing, Multiple Access
CSMA/CD	CSMA with Collision Detection
CSS	Communications Support System
CUDIXS	Common User Digital Information Exchange System
CVSD	Continuous Variable Slope Delta
CW	Continuous Wave
DAA	Designated Accreditation Authority
DAMA	Demand Assigned Multiple Access
DAR	Distortion Adaptive Receiver
dB	Decibels
DBB	Direct Broadcasting Band
DBMS	Database Management System
DC	Direct Current
DCS	Defense Communications System
DDN	Defense Data Network
DES	Digital Encryption Standard
DFT	Discrete Fourier Transform
DISA	Defense Information Services Agency
DM	Delta Modulation
DMA	Direct Memory Access
DoD	Department of Defense
DPCM	Differential Pulse Code Modulation
DPSK	Differentially Encoded Phase Shift Keying
DSC	Digital Selective Calling
DSCS	Defense Satellite Communications System
DSP	Digital Signal Processing
DTD	Data Transport Device
ECM	Electronic Counter Measures
EHF	Extremely High Frequency
ELF	Extremely Low Frequency

ES-IS	Easy System-to-Intermediate System
ET	Exchange Termination
EUROCOM	European Command
FCC	Federal Communications Commission
FDDI	Fiber Distributed Data Interface
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correcting
FEK	Frequency Exchange Keying
FEP	FLTSAT EHF Package
FIST	Fleet Intelligence Support Terminal
FIR	Finite Impulse Response
FLTBCST	Fleet Broadcast
FLTSATCOM	Fleet Satellite Communications
FM	Frequency Modulation
FO	Follow-On
FOSIC/F	Fleet Ocean Surveillance Information Center/Facility
FOTC	Force OTH-T Track Coordinator
FSK	Frequency Shift Keying
FTAM	File Transfer, Access, and Management
FTS	Federal Telephone System
Gbps	Giga-bits per second
GENSER	General Service
GEO	Geostationary Orbiting
GFCP	Generic Front-End Communications Processor
GHz	Giga-Hertz
GLOBIXS	Global Information Exchange System
GMDSS	Global Maritime Distress and Safety System
GOSIP	Government Open Systems Interconnection Profile
GOTS	Government Off-the-Shelf
GPS	Global Positioning System
GTE	General Telephone and Electric Company
GUI	Graphic User Interface
HCDM	Hybrid Companding Delta Modulation
HF	High Frequency
Hz	Hertz
IEEE	Institute for Electrical and Electronics Engineers
IF	Intermediate Frequency
IIR	Infinite Impulse Response
IMO	International Maritime Organization
INFOSEC	Information Security
Inmarsat	International Maritime Satellite
INTELSAT	Intelligence Satellite
IOC	Initial Operating Capability

IP	Internet Protocol
IPC	Interprocess Communications
ISB	Independent Side Band
ISC	Information Systems Center
ISDN	Integrated Services Digital Network
ISO	International Standards Organization
JOTS	Joint Operational Tactical System
JTF	Joint Task Force
kbps	Kilobits per second
kHz	Kilohertz
Km	Kilometers
KMP	Key Management Plan
LAN	Local Area Network
LDM	Linear Delta Modulation
LEASAT	Leased Satellite
LED	Light Emitting Diode
LEO	Low Earth Orbiting
LF	Low Frequency
LLC	Logical Link Control
LOS	Line-of-Sight
LPC	Linear Predictive Coding
LPD	Low Probability of Detection
LPI	Low Probability of Intercept
LSB	Lower Side Band
LT	Loop/Line Termination
MAC	Medium Access Control
MAN	Metropolitan Area Network
MARISAT	Maritime Satellite
Mbps	Megabits per second
MF	Medium Frequency
MFSK	Multiple FSK
MHS	Message Handling Service
MHz	Megahertz
MIL-STD	Military Standard
MLS	Multi-Level Security
MM	Millimeter
MOA	Memorandum of Agreement
MSD	Modular Security Device
MSK	Minimum Shift Keying
MTF	Message Text Format
MUX	Multiplexer
NASA	National Aeronautics and Space Administration

NATO	North Atlantic Treaty Organization
NAVCAMS	Naval Communications Area Master Station
NAVCOMPARS	Naval Communications Processing and Routing System
NAVMACS	Navy Modular Automated Communications System
NAVSTAR	Navigation Satellite Timing and Ranging
NBFM	Narrow Band FM
NESP	Navy EHF Satellite Program
NETSEC	Network Security
NGC	Next Generation Cutter
NGCR	Next Generation Computer Resources
NIST	National Institute of Standards and Technology
NRL	Naval Research Laboratory
NSA	National Security Agency
NT	Network Terminal or Network Termination
NTCS-A	Navy Tactical Command System-Afloat
NTDS	Navy Tactical Data System
NWP	Naval Warfare Publication
ODA	Office Document Architecture
OOK	On-Off Keying
OPNOTE	Operator Note
OPORD	Operations Order
OPSEC	Operational Security
OPTASK	Operations Task
OQPSK	Offset Quadrature Phase Shift Keying
OS	Operating System
OSI	Open Systems Interconnect
OTCIXS	Officer-in-Tactical Command Information Exchange System
OTH-T	Over-the-Horizon Targeting
PBX	Private Branch Exchange
PCM	Pulse-Code Modulation
PDM	Pulse Duration Modulation
PHY	Physical Layer Protocol
PLM	Pulse Length Modulation
PM	Physical Medium
PMD	Physical Media Dependent
PSK	Phase Shift Keying
PWN	Pulse Width Modulation
QPSK	Quadrature Phase Shift Keying
RC	Resistance-Capacitance
RCA	Radio Corporation of America
RDF	Radio Direction Finding
RF	Radio Frequency
RFI	Radio Frequency Interference
RI	Routing Indicators

RIC	Resistance-Induction-Capacitance
S/N	Signal to Noise Ratio
SAFENET	Survivable Adaptable Fiber Optic Embedded Network
SAR	Search and Rescue
SART	Search and Rescue Transponder
SATCOM	Satellite Communication
SCCN	Secure Command and Control Network
SCSI	Small Computer System Interface
SDH	Synchronous Digital Hierarchy
SDN	Secure Data Network
SECVOX	Secure Voice
SES	Ship Earth Station
SHF	Super high Frequency
SITOR	Simplex Teleprinting Over Radio
SMT	Station Management
SNAcP	Subnet Access Protocol
SNDCP	Subnet Dependent Convergence Protocol
SNICP	Subnet Independent Convergence Protocol
SOE	Standard Operating Environment
SOLAS	Safety of Life at Sea
SONET	Synchronous Optical Network
SPAWAR	Space and Naval Warfare Systems Command
SPE	Synchronous Payload Envelope
SPK	Single Point Keying
SQL	Structured Query Language
SQNR	Signal-to-Quantization Noise Ratio
SRAM	Static Random Access Memory
SSB	Single Side Band
SSBLT	Source Synchronized Block Transfer
SSBN	Source Synchronized Ballistic Missile Submarine (Nuclear)
SSIXS	Submarine Information Exchange Subsystem
STD	Standard
STT	Shore Targeting Terminal
STS	Synchronous Transport Signal
STU	Secure Telephone Unit
STW	Strike Warfare
SUBOPAUTH	Submarine Operating Authority
SVADM	Song Mode Voice Digital Adaptive Delta Modulation
TA	Terminal Adapter
TACINTEL	Tactical Intelligence
TADIXS	Tactical Data Information Exchange System or Tactical Information Exchange System
TASM	Tomahawk Anti-Surface Missile
TC	Transmission Convergence
TCC	Tactical Command Center

TCP/IP	Transmission Control Protocol/Internet Protocol
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TDP	Tactical Data Processor
TE	Terminal Equipment
TELSAT	Telecommunications Satellite
TFS	Traffic Flow Security
THz	Tera Hertz
TLAM	Tomahawk Land Attack Missile
TOR	Teleprinting Over Radio
TP2	Transport Protocol 2
TRANSEC	Transmission Security
TRAP	Tactical and Related Application
TRE	Tactical Receive Equipment
TTY	Teletype
TV	Television
T/VA	Threat/Vulnerability Analysis
TWCS	Tomahawk Weapon Control System
UFO	UHF Follow-On
UHF	Ultrahigh Frequency
UPP	User Partnership Program
USB	Upper Side Band
UTC	Coordinated Universal Time
UV	Ultraviolet
VC	Virtual Channels
VCI	Virtual Channel Identifier
VF	Voice Frequency
VHF	Very High Frequency
VHSIC	Very High Speed Integrated Circuits
VLF	Very Low Frequency
VLSI	Very Large Scale Integration
VME	Versa Module Europe
VPI	Virtual Path Identifier
VSAT	Very Small Aperture Terminal
VT	Virtual Terminal
XTP	Express Transport Protocol

APPENDIX B REFERENCED DOCUMENTS

IEEE 1003.1	Portable Operating System Interface for Computer Environments (POSIX), 1988
IEEE 1003.4	Draft Standard for Information Technology - System Application Program Interface - POSIX Real-time Extension [C Language], Draft 12, February 1992
IEEE 1003.13	Draft Standard for Information Technology - Standard Application Environment Profile - POSIX Real-time Application Support (AEP), Draft 5, February 1992
IEEE 802.3	Ethernet Local Area Network (LAN), A Bus Utilizing Carrier Sense, Multiple Access With Collision Detection (CSDA/CD) As the Access Method
ISO 8802-3	Information Processing Systems - Local Are Networks - Part 3: Carrier Sense Multiple Access with Collision Detection (CDMA/CD) Access Method and Physical Layer Specifications, September 1990.
ISO 8473	Information Processing Systems - Data Communications - Protocol for Providing the Connectionless-Mode Network Service, December 1988
ISO 9542	Information Processing Systems - Telecommunications and Information Exchange Between Systems - End User to Intermediate System Routing Exchange Protocol for Use in Conjunction with the Protocol for Providing the Connectionless-mode Network Service, August 1988.
ISO 8073	Information Processing Systems - Open Systems Interconnection - Connection Oriented Transport Protocol Specification, December 1988
ISO 8648	Information Processing Systems - Open Systems Interconnection - Internal Organization of the Network Layer, February 1988
DIS 10789	Information Processing Systems - Intermediate System to Intermediate System Routing Protocol, January 1991

MIL-STD-188-110A114A	Electrical Characteristics of Digital Interface Circuits, 30 September 1985
MIL-HDBK-00360035	Survivable Adaptable Fiber Optic Embedded Network II (SAFENET), Draft, 15 January 1991
	A Structured Overview of Digital Communications – A Tutorial Review-Part II, Bernard Sklar, Institute for Electrical and Electronic Engineers (IEEE) Communications, 1983.
	The Application of NAVISTAR Differential GPS in the Civilian Community", Beser and Parkinson's *Navigation,* Volume 29, 1982.