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TELEPHONE PRIMER

by

Patrick J. Hovatter

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The telephone is the most common and widely used electronic communication device in use today. Because of this, the military is heavily reliant on the telephone system and considers it the primary voice communications medium. Furthermore, recent technological advances will dramatically change the telephone as we know it today. This thesis will take a comprehensive look at the telephone in today's complex telecommunications environment. It will describe the technical aspects of individual components as well as how the system works as a whole. The divestiture of AT&T will be analyzed, especially the effects it has had on the military. After describing the historical and technical aspects of the telephone system, the thesis will focus its attention on military telephone programs and upgrades being planned to increase telephone capabilities and survivability.

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Telephone Primer

by

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ABSTRACT

The telephone is the most common and widely used electronic communication device in use today. Because of this, the military is heavily reliant on the telephone system and considers it the primary voice communications medium. Furthermore, recent technological advances will dramatically change the telephone as we know it today. This thesis will take a comprehensive look at the telephone in today's complex telecommunications environment. It will describe the technical aspects of individual components as well as how the system works as a whole. The divestiture of AT&T will be analyzed, especially the effects it has had on the military. After describing the historical and technical aspects of the telephone system, the thesis will focus its attention on military telephone programs and upgrades being planned to increase telephone capabilities and survivability.

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ACRONYMS

acronym	*definition*
ACOC	Area Communications Operations Center
ADPCM	Adaptive Differential Pulse Code Modulation
AM	Amplitude Modulation
AMPS	Advanced Mobile Phone Service
ANDVT	Advanced Narrowband Digital Voice Terminal
APD	Avalanche Photo Detector
ASD	Access Security Device
ASK	Amplitude Shift Keying
AUTOSEVOCOM	Automatic Secure Voice Communications network
AUTOVON	Automatic Voice Network
BER	Bit Error Rate
B-ISDN	Broadband Integrated Services Digital Network
BITS	Base Information Transfer System
BOC	Bell Operating Company
BORSCHT	Battery, Overvoltage, Ringing, Signaling, Coding, Hybrid, and Test
BRI	Basic Rate Interface
CC	Call Controller
CCITT	International Telegraph and Telephone Consultive Committee
CCS	Common Channel Signaling
CCS	Hundred Call Seconds
CIK	Crypto Ignition Key
CLASS	Custom Local Area Signaling Service
CNS	Commercial Network Survivability
CO	Central Office
CODEC	Coder-Decoder
CPE	Customer Premise Equipment
CSI	Commercial SATCOM Interconnectivity
CVSD	Continuously Varying Slope Delta modulation

DACS	Digital Access and Cross-connect System
DAMA	Demand Assigned Multiple Access
DCOSS	Defense Communications Operations Support System
DCS	Digital Cross-connect System
DCTN	Defense Commercial Telephone Network
DECCO	Defense Commercial Communications Office
DMATS	Defense Metropolitan Administrative Telephone System
DPA	Demarcation Point Annex
DPAS	Digital Patch and Access System
DPCM	Differential Pulse Code Modulation
DS	Digital Service
DSB	Double Side Band
DSCS	Defense Satellite Communications System
DSN	Defense Switched Network
DTSW	Defense Telephone System Washington
EAS	Extended Area Service
EHF	Extremely High Frequency
EIRP	Equivalent Isotropically Radiated Power
EMI	Electro Magnetic Interference
EO	End Office
ESF	Extended Super Frame
ESS	Electronic Switching System
ETS	European Telephone System
FCC	Federal Communications Commission
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FM	Frequency Modulation
FSK	Frequency Shift Keying
FTS	Federal Telecommunications System
FX	Foreign Exchange
GPS	Global Positioning System
GRIN	Graded Index multimode

GTOC	General Telephone Operating Company
HDTV	High Definition Television
ICC	Interstate Commerce Commission
IDCS	Integrated Defense Communications System
IDS	Integrated Data Services
IF	Intermediate Frequency
ILD	Injector Laser Diode
IMTS	Improved Mobile Telephone Service
ISDN	Integrated Services Digital Network
ISO	International Standards Organization
ITT	International Telephone and Telegraph
ITU	International Telecommunications Union
IVSN	Initial Voice Switched Network
IXC	Inter-exchange Carrier
JRSC	Jam Resistant Secure Communications System
JTU	Japan Telephone Upgrade
KTS	Key Telephone Set
KTU	Korean Telephone Upgrade
LAN	Local Area Network
LATA	Local Access Transport Area
LEC	Local Exchange Carrier
LED	Light Emitting Diode
LOS	Line Of Sight
MCA	Maximum Calling Area
MCI	Microwave Communications Incorporated
MDF	Main Distribution Frame
MER	Most Economical Routing
MF	Multi-Frequency
MFJ	Modified Final Judgment
MLPP	Multi-Level Precedence and Preemption
MTS	Military Telephone System
MTSO	Mobile Telephone Switching Office

MTT	Mobile Transportable Telecommunications
NDCCA	Navy Data Communications Control Architecture
NCS	National Communications System
NETS	National Emergency Telecommunications System
NLP	National Level Program
NMAC	NETS Maintenance and Administration Center
NMCC	National Military Command Center
NPA	Numbering Plan Areas
NRZ	Non-Return to Zero
NSEP	National Security Emergency Preparedness
NT	Network Terminator
NTIA	National Telecommunications Information Administration
OCC	Other Common Carrier
O&M	Operations & Maintenance
OOK	On-Off Keying
OSI	Open System Interconnection reference model
PAM	Pulse Amplitude Modulation
PAT	Precedence Access Thresholding
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PIN	Positive Intrinsic Negative
PLL	Phase Locked Loop
PM	Phase Modulation
POP	Point Of Presence
PPM	Pulse Position Modulation
PRI	Primary Rate Interface
PSK	Phase Shift Keying
PSN	Public Switched Network
PTN	Public Telephone Network
PTT	Foreign Post, Telephone, and Telegraph Exchange
PUC	Public Utilities Commission
PWM	Pulse Width Modulation

RBOC	Regional Bell Operating Company
RCC	Radio Common Carrier
REN	Ring Equivalence Number
RF	Radio Frequency
ROR	Rate Of Return
RSP	Red Switch Project
RSU	Remote Switching Unit
RZ	Return to Zero
SAT	Supervisory Audio Tone
SATCOM	Satellite Communications
SCP	Secure Conferencing Program
SCP	Signal Control Point
SDM	Space Division Multiplexing
SHF	Super High Frequency
SMSA	Standard Metropolitan Statistical Area
S/N	Signal to Noise Ratio
SNI	Standard Network Interface
SRCF	Sub Regional Control Facility
SSP	Service Switching Point
ST	Signaling Tone
STP	Signal Transfer Point
STU	Secure Telephone Unit
SVIP	Secure Voice Improvement Program
SVS	Secure Voice System
SXS	Step-by-Step Switch
TA	Terminal Adaptor
TASI	Time Assignment Speech Interpolation
TAT	TransAtlantic Telephone Cable
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TE	Terminal Equipment
TELCO	Telephone Company

TELSEC	Telephone Security
TT&C	Telemetry, Tracking, and Control
VAC	Value Added Carrier
VQC	Vector Quantizing Code
WATS	Wide Area Telephone Service
WDM	Wavelength Division Multiplexing
WNDP	Worldwide Numbering and Dialing Plan

I. INTRODUCTION

The field of telecommunications is in a state of revolution. At the forefront of this revolution are the changes that are taking place in today's telephone system. For nearly 100 years, Americans were blessed with the best telephone system in the world thanks to American Telephone and Telegraph (AT&T). But for several reasons, the monopoly AT&T created and that had been so long protected by the government was split up by the government. Coupled with this split were technological advances in optical communications and digital electronics. Without a doubt, the telephone has become the world's most relied upon means of electronic communication. Reliance upon the telephone is so high that even the Department of Defense (DOD) considers the telephone its primary means of communication whether in time of crisis or in peacetime. Even though the telephone is so important, no textbook offers a full treatment on the background of the defense aspects of telephony. For this reason, this thesis will attempt to cover all aspects of the telephone from early telephone history through military telephone programs.

The thesis is arranged in a logical fashion to provide background for each successive chapter. The thesis does not get too in-depth in any one area, but enables the reader to gain enough knowledge on each subject area to be conversant and understand basic concepts without having to use excessive amounts of reference material.

Therefore, to set the stage, a brief history of the telephone is presented in Chapte Two. Some of the historical aspects are merely for the curious, but the big picture that the chapter details is important to understand in order to comprehend why our telephone system is set up the way it is today. Chapter Three concerns itself with regulation and is more or less a continuation of Chapter Two in that it is historically based. However, Chapter Three provides insight into just how detailed the regulation is and how telephone boundaries are established. Regulation has essentially dictated the architecture of today's telephone network.

In Chapter Four, the basic technologies are explained. In the first few sections of the chapter, the differences between analog and digital signals are discussed as are the more common methods of transmission. These were included to give the reader an understanding of what the remainder of the thesis would discuss.

Chapter Five provides a picture of where telephony is heading, the Integrated Services Digital Network (ISDN). Although ISDN is the key to the future, there is a great deal of debate over the merits of ISDN due to standardization and other concerns. In either case, it is obvious that we are heading in the direction of an integrated network at least very similar to ISDN. Even the military is selecting ISDN as the network standard of choice.

Finally, Chapter Six wraps up the thesis by presenting the military aspects of the telephone. Chapter Six primarily concerns itself with the Defense Switched Network (DSN), since DSN will virtually encompass all other telephone upgrades. However, secure telephones and survivability programs are also discussed.

It is important to emphasize the reliance we all put on the telephone. Furthermore, the telephone as we know it today probably will not be anything close to the planned system. There are several companies pushing to combine the telephone and computer into a "do it all" workstation, including a videophone. The technology is all there; all that is needed is the opportunity to make the investment worthwhile. An all-digital broadband fiber optics network will most likely be the solution.

II. HISTORY OF THE TELEPHONE

A. BACKGROUND

The telephone as we know it today is the product of over 100 years of devotion and research by many scientists from around the world. As early as 1854, a Frenchman by the name of Charles Bourseul published an article describing a device he believed would electrically transmit speech. Other notable scientists of this era who were working to achieve electronically transmitted speech were Italy's Antonio Meucci, Germany's Philipp Reis, and America's Alexander Bell. It was through relentless pursuit of this concept, and a bit of chance, that Alexander Bell was soon to prove his theory.

B. ALEXANDER G. BELL INVENTS THE TELEPHONE

Alexander Graham Bell was born on March 3, 1847, in Edinburgh, Scotland. His father and grandfather who were teachers specialized in correction of defective speech, and his father was the inventor of sign language, still used today to communicate with the deaf and dumb. Naturally, with such a prestigious background, the younger Bell followed suit and began to teach deaf children.

In 1868, the elder Bell visited Boston to demonstrate his teaching methods and made an impression with Sarah Fuller, the principal of a school for the deaf. Two years later, the elder Bell moved his family to Canada and planned another trip to Boston. When Miss Fuller learned that the elder Bell would be visiting Boston in October of that year, she requested he conduct a course in sign language for her school. The elder Bell was unable to attend, but instead wrote Miss Fuller acknowledging his son's ability in this area and recommending him. Based on this recommendation, Miss Fuller persuaded the Boston Board of Education to hire Alexander Bell as a full time teacher. On April 1, 1871, Bell accepted the position and moved to Boston. It was through his teachings that he would meet Gardiner Hubbard and Thomas Sanders who would provide the necessary encouragement and financial backing which lead to the invention of the telephone. A gentlemen's agreement between Bell, Hubbard, and Sanders was reached in October 1874, where each person would receive one third interest from any patents that would result from Bell's research. It was at this time that Thomas Watson was chosen to be Bell's invaluable laboratory assistant. [Ref. 1: p. 5-9]

Several factors lead to the invention of the first telephone. First, Bell learned of experiments conducted by Germany's renowned scientist Helmholtz that involved electromagnetic tuning forks. Since Helmholtz's works were only published in German, and Bell was not fluent in German, Bell misinterpreted the experiments believing Helmholtz had transmitted vowel sounds electronically. This convinced Bell that speech transmission was possible. Secondly, Bell believed that undulatory current was needed to carry the information contained in speech. Undulatory current is a continuous signal that varies in strength or intensity representing the way sound waves travel through air. [Ref. 1: p.7]

Shortly thereafter, Watson and Bell set out to make the "harp telephone," which was a series of steel rods that would vibrate near an electromagnet and a permanent magnet. The idea quickly became too complicated and was discarded. However, this concept gave birth to the use of a single reed that would be vibrated by a stretched membrane which was displaced by the speaker's voice moving the air in front of it. Bell got the idea of the stretched membrane from a invention called a phonautogragh that was used to teach deaf students. It occurred to Bell that the human ear worked on a similar principle, and he decided to consult Dr. Clarence Blake, a noted Boston aurist. Dr. Blake provided Bell with bones from a human ear which Bell experimented with and concluded that the thin membrane moved by sound could move a heavier object and make it vibrate accordingly. Furthermore, a larger and heavier membrane would even be able to move a metal rod. [Ref. 1: p. 10-11]

In early 1875, Bell was becoming fatigued due to the stress of inventing and teaching. It was Joseph Henry, Secretary of the Smithsonian Institution, that provided Bell with the encouragement and direction to keep pursuing his goal. Shortly after this, Bell and Watson were experimenting with their harmonic telephone when they discovered that the steel reeds they had been using were adjusted too tightly and prevented proper vibration. By setting the proper tension, the reeds vibrated across the pole of the magnet producing the undulatory current Bell sought. For the first time in history, an electrical current carried an audio signal. This lead quickly to Bell combining what he had learned from the stretched membrane and the harmonic telephone into one device. On June 2, 1875, Watson heard Bell's voice over their instrument, which gave birth to the telephone. [Ref. 1: p.12-14]

After his success, Bell began to draft the specifications for his patent application. Several delays were experienced as Bell was pursuing simultaneous patents in the United States and United Kingdom. Finally, in February 1876, Bell filed his patents. Unexpectedly, another inventor, Elisha Gray, filed a similar patent only hours after Bell. There was an important difference in that Gray's patent application was for a "caveat patent," which means the person filing the patent only intends to invent or process that device. Obviously, Bell's patent won out, and even Gray gave credit to Bell for the invention of the telephone. [Ref. 1: p.14] An interesting side note is that Bell offered to sell his patent to Western Union for $100,000, but Western Union declined, stating it was only interested in telegraphs [Ref. 2: p. 15].

Since the first telephone's voice quality was very poor, Bell continued working to improve this aspect. His work lead him to the so called "liquid" transmitter. The transmitter was basically a slender platinum rod suspended in a mild acidic solution. The purpose of the acidic solution was to complete an electric circuit. The rod being suspended by the diaphragm would be moved up and down by the speaker's voice, thus causing the undulatory current to flow. [Ref. 1: p. 15]

Finally, on March 10, 1876, Bell and Watson were ready to conduct the final test. Watson was in another room with the receiver, while Bell was filling the transmitter with the acidic solution. The irony came when Bell spoke the infamous words, "Mr. Watson, come here; I want you!" Watson had clearly heard the words over his receiver, but what Bell really wanted was for Watson to help him clean off the acid he had spilled all over his trousers. Once Watson informed Bell the apparatus worked, the trousers were quickly forgotten. [Ref. 1: p. 15-16]

C. ORIGINS OF THE BELL TELEPHONE COMPANY

Although the invention was considered to be truly great, the scientific community and the general public were not aroused, probably because of lack of foresight and knowledge. It was at the Philadelphia Centennial Exposition that Bell got the endorsement he needed. Quite coincidentally, the Emperor of Brazil, Dom Pedro, who had met Bell previously, was in attendance. After recognizing Bell, he asked for a demonstration for himself and a team of judges that included Lord Kelvin. The demonstration was a tremendous success. As a result, the scientific community gave Bell the endorsement he needed so that the public would accept it as a useful invention. [Ref. 1: p. 16-18]

In order to get public support, financial backing was required in excess of what the three entrepreneurs could muster. Sanders alone had invested over $110,000 at this point. Also at this time, Watson became the fourth partner, but he had little financial resources to invest. So the four men formally entered into an agreement recorded by a deed of trust dated July 9, 1877. The deed created the Bell Telephone Company, named Hubbard as the trustee, and provided the financial resources through the sale of stock. [Ref. 1: p. 20]

The first decision Hubbard made was to lease, not sell, the telephones. This decision further reflected the policy that anyone developing a telephone system throughout the country would be licensed to operate under the Bell patents but would not be allowed to own the equipment. In essence, this decision set the stage for company policy for the next 100 years. Furthermore, by deciding against sales, the emphasis could be placed on performance and attaining a universal standard of service.

In 1877, there were about 100 telephones in use which was not providing enough revenue to sustain the business. Therefore, the time had come to solicit additional investors. Sanders, being the more financially minded, sought some of his friends to invest. The investors, all of them from New England, required that development be confined to the New England area. Accordingly, the New England Telephone Company was formed on February 12, 1878.

By 1878, there were roughly 800 telephones in use nationwide, and they were only sold in pairs [Ref. 3: p. 211]. The reason telephones were sold in pairs was that there were no exchanges or switches at that time. There was no network, just one wire linking only two telephones. Bell foresaw the need for exchanges, but considered it impractical for the time being. Meanwhile, George Coy of New Haven, Connecticut, proved Bell wrong. He connected eight lines and 21 telephones to a crude and primitive switchboard. The installation of these exchanges grew rapidly and was virtually being used in cities coast to coast by 1881.

The outlook for success in the New England Telephone Company was excellent and encouraged the associates to expand their business to the rest of the country. Therefore, the Bell Telephone Company was rechartered as a separate entity from the New England Telephone Company under Massachusetts law on June 29, 1878.

D. THEODORE VAIL HIRED TO RUN BELL TELEPHONE

By this time, the Bell Telephone Company had established its presence. Its main competitor, Western Union, soon realized its earlier error not to buy out Bell's patents, and now had to pay $2 million to secure similar patents from Elisha Gray and Thomas Edison. Much of Western Union's prowess came from the acquisition of the Edison microphone. The competition continued to grow as Western Union's subsidiary, the American Speaking Telephone Company, had established a customer base of over 50,000 subscribers in over 55 cities. Furthermore, Western Union owned Western Electric which manufactured the best telephone and telegraph equipment in the country and held numerous important patents. Bell Telephone in comparison was much smaller, but gaining ground. As the competition continued, the battle grew bitter. [Refs. 1: p. 21-26, 2: p. 15-16]

On the surface, Bell Telephone versus Western Union looked like a David and Goliath mismatch. Western Union was one of the largest companies in America, with stable financing, existing rights of way stemming from the telegraph business, and a well known manufacturing subsidiary. But Hubbard knew aggressive, young Theodore Vail, then the General Superintendent of the United States Railway Mail Service. Hubbard convinced Vail to hire on to the fledgling Bell Telephone Company. Vail's first action was to aggressively take on Western Union.

Vail took Western Union to court on the grounds of patent infringement. He further rejected Western Union's proposal to sell out to them or negotiate the sale of Bell's long distance service. In fact, Vail took the offensive by threatening to enter the telegraph market. For many, the decision of the court was a surprise. On November 10, 1979, the court ruled in favor of Bell Telephone. Western Union, once the Goliath of the industry, was now engaged in a strategic retreat. It agreed to withdraw from the telephone business and sell its patents and the American Speaking Telephone Company to Bell for $300,000. In return, Bell agreed to pay Western Union 20 percent of future royalties for a specified period and to stay out of the telegraph business. [Ref. 2 : p. 16]

The national scope of the telephone company was made official when the Bell Telephone Company became the National Bell Telephone Company on February 17, 1879. This occurred during the Western Union battle. This act combined the assets and responsibilities of the Bell Telephone Company and the New England Telephone Company. Financial backing was prompted by William Forbes, who also got his friends to invest. Due to his overwhelming financial investments, Forbes was named president with Vail as general manager. Forbes' and Vail's leadership qualities provided a synergism in management, at least for a while. [Ref. 1: p. 26]

Soon after incorporation, it became evident more funding would be required to bring about true universal service. Funding was again provided by the Forbes investment group and by passage of a special act in the Massachusetts legislature. This action created the American Bell Telephone Company on March 20, 1880, with Forbes and Vail at the helm. The new company immediately formed centralized staffs to provide the licensed companies advisory services on matters related to engineering, legal, financial, and other services. The pursuit of a manufacturing branch was solved in 1882, when the Western Electric company was purchased. Also, the first steps towards establishing the famous Bell Laboratory were made, which at that time consisted of only a very tired Thomas Watson. [Ref. 1: p. 27-28]

In the 1880s, Vail aggressively fought off competition and filed over 600 lawsuits. His goal was to control the industry with or without patents and to provide universal service nationwide. Meanwhile, Bell, the inventor of the telephone, was tired of the financial dealings and had developed interests elsewhere. Gradually, Bell withdrew himself from the telephone company he created, and by 1881 had completely withdrawn. His only interest in the company was as a minority stockholder, although he did appear on behalf of the company during several lawsuits. [Ref. 3: p. 211]

E. AMERICAN TELEPHONE AND TELEGRAPH FORMED

As the American Bell Telephone Company exercised its strategy to expand its long lines, it once again required an infusion of capital. However, Massachusetts law prevented the infusion of the amount of capital the company requested. Therefore, a decision was made to create the American Telephone and Telegraph (AT&T) under New York law. Vail became the first president of AT&T on February 28, 1885. The company declared its purpose was to extend telephone service to every state, city, and even to Canada and Mexico. AT&T, as a subsidiary of the American Bell Telephone Company, would operate the long distance lines to connect the local exchanges of the Bell companies. By 1889, long distance service was installed from Buffalo, New York, on the west, to Washington, D.C. in the south. Long distance lines coast to coast would not come until 1915 when a line was strung from New York to San Francisco. [Ref. 4: p. 9]

Unfortunately, by 1887 growing differences between Forbes and Vail caused Vail to resign. Vail was the ultimate expansionist and believed that capital should be retained in the company to support growth, especially considering that most of the patents would expire in six years. On the other hand, Forbes saw a need to pay a premium dividend to those who invested. However, before his resignation, Vail accomplished setting in motion the basic elements to ensure American Bell's future. Vail did this through vertically integrating internal manufacturing, franchising local exchanges that were required to give Bell an equity position, and providing long distance lines to connect the local exchanges, thus creating an efficient network. [Ref. 3: p. 212]

The years following Vail's resignation introduced more growth for American Bell, but also independent telephone companies (TELCOs) were sprouting up everywhere encouraged by the public's view of the monopolistic American Bell Telephone Company. American Bell Telephone Company retaliated by refusing to allow the independent

TELCOs to connect with Bell's long distance network or buy equipment from Bell-owned Western Electric.

In an effort to stave off competition, American Bell once again went to its financiers for assistance. Since Massachusetts law precluded another infusion of capital due to a law limiting the assets of a company, the solution was achieved by shifting all of American Bell's assets under AT&T in accordance with New York law. The transfer of assets was completed in December of 1889 and transformed AT&T into the parent company, with American Bell and Western Electric as subsidiaries. Even with this large influx of capital, AT&T lacked the dynamic leadership of Vail and consequently was missing some key points. One of these points was automatic switching. Since the independent TELCOs were particularly strong in rural areas, they invented and installed an automatic switch that the subscriber could control by dialing the numbers himself. Bell, for whatever reason, thought customers would not like this feature, and used manual switches that required operator intervention to dial. After realizing the significance of its error, Bell found itself licensing switching technology from the independent TELCOs. Although AT&T continued to grow, the independent TELCOs' combined efforts accounted for 50 percent of the six million telephones in service in 1907. [Refs. 2: p.17, 3: p. 213]

F. AMERICAN TELEPHONE AND TELEGRAPH CHANGES STRATEGY

Fortunately for AT&T, the Boston based group headed by the conservative Forbes was replaced by a Wall Street syndicate headed by J. P. Morgan who promptly and wisely returned Vail to be president of AT&T. Together the two men shared the same philosophy and believed AT&T should be a single force in the telephone industry. Vail used the term "universality," which meant he was admitting to the concept of a public monopoly. Determined to achieve this concept, Vail intensified his campaign against the independent TELCOs by lowering rates to undercut them and buying them out whenever possible. Buy outs were made considerably easier since J. P. Morgan controlled a major portion of the independent TELCOs' financial resources and would clamp down on these resources to make them vulnerable to takeover. Additionally, Vail reorganized Western Electric by laying off 12,000 employees and even started selling his telephones to the independent TELCOs. This latter move appeased the TELCOs while AT&T reaped the profits. [Refs. 2: p. 21, 3: p. 214]

Vail continued his aggressive strategy with the purchase of 30 percent of Western Union in 1909 which gave AT&T control of the company. In 1910 Vail became president of Western Union as well as AT&T. This move raised additional questions concerning Vail's concept of universality.

The way Vail viewed the situation regarding corporate strategy, he had three options: competition, "postalization," and regulation. He eliminated competition believing that the telephone was a natural monopoly and competition would only bring inefficient redundancy. "Postalization" was an idea held by many since the United States Postal Service had developed an interest in telephones in earlier years after operating the first telegraph service in the United States. Even the Postmaster General himself favored combining the telephone and mail business. However, Vail rejected this idea, regarding it as flawed due to lack of accountability in a government-owned monopoly. Vail's decision was set forth in the 1910 annual report. [Ref. 2: p. 26]

Vail's response was to shift his strategy to publicly selling the idea of a regulated monopoly. He argued that the telephone industry was a natural monopoly and that the public would gain through less competition and more economies of scale. He further called for a regulatory commission similar to that of the railway regulatory commission and even actively lobbied to get stronger regulation. Vail believed that universality could not be achieved by any other means except a monopoly which should be privately owned and government controlled. [Ref. 4: p. 10]

As Vail continued his fight against the independent TELCOs, the TELCOs fought back by using state antitrust laws. This lead to the involvement of the Interstate Commerce Commission (ICC) which had limited jurisdiction over interstate telephone business. The ICC opened an investigation concerning AT&T's attempts to monopolize the market which eventually brought AT&T to the bargaining table. [Ref. 4: p. 11]

G. THE KINGSBURY COMMITMENT

AT&T soon recognized that the trustbusters were not convinced of the need for a regulated monopoly. In just two months, Woodrow Wilson would be President of the United States, and he had made antitrust legislation a major issue in his campaign. Vail sent his Vice President, Nathan Kingsbury, to work out an agreement with the Wilson administration. In December 1913, the Kingsbury Commitment was reached. Its terms specified that AT&T was to sell its interest in Western Union, purchase no more independent TELCOs without Interstate Commerce Commission (ICC) approval, and allow the existing independent TELCOs connection to the Bell system. This commitment defeated the first of many anti-trust suits that would follow in the years to come. Also, it enabled the strategy of AT&T to withdraw from markets that went head-to-head with the independent TELCOs and strengthened Vail's belief that universality could only be achieved through a government-controlled monopoly. However, this commitment allowed the growth of several large independent TELCOs in their respective geographical areas. [Refs. 3: p. 214, 4: p. 11]

Also during this time frame, Vail recognized the growth of technology in the field of radio and vacuum tubes that threatened the telephone's wire technology. Therefore, Vail enlarged the research and development effort to include basic technology which had not been done up until that time. This research outfit, which would eventually become the Bell Labs in 1924, was so successful that by 1920 it possessed nearly all patents associated with radio (Bell Labs was created from a merger of Western Electric staff and AT&T Headquarters staff). [Ref. 2: p. 18]

Relations between the independent TELCOs and AT&T were strengthened in 1912 when AT&T promised not to back out of the Kingsbury commitment and to work cooperatively with the independent TELCOs. Soon both entities realized that they could rely on state and federal regulation to guarantee them a fair rate of return (ROR), and that regulation imposed rigid entry barriers that were nearly impossible to overcome for any new competitors. Furthermore, Bell's share of the market was now stabilized at approximately 75 percent. [Ref. 4: p. 12]

However, the advance of technology and the superb research done by Bell Labs soon landed them back at the bargaining table through an odd sort of events. It seems that although Bell Labs owned most of the radio patents, General Electric and Westinghouse also had considerable expertise and patents in this area. The resulting complications to avoid patent infringements among the three companies forced General Electric to consider selling its expertise to a foreign firm, American Marconi. This got the attention of the Department of the Navy which feared foreign control of communications devices for naval use. The Navy asked General Electric to consolidate its patents with other American firms. Out of this, AT&T, General Electric, and Westinghouse formed a new company called Radio Corporation of America (RCA).

Armed with the complete set of radio and vacuum tube technology, Bell diversified into broadcasting. The competing broadcasting stations were denied access to the Bell facilities and sought to lease Western Union facilities. The problem was that Western Union's lines were not of voice grade and were not as extensive. Finally, in 1926, AT&T agreed to withdraw from the broadcasting business to avoid lawsuits and sold its chain of stations to RCA whose affiliate, the National Broadcasting Company (NBC), agreed to lease toll telephone facilities from AT&T. [Ref. 2: p. 18-20]

In other developments, the Bell Telephone system opened its first transcontinental line on January 25, 1915, connecting New York and San Francisco. Shortly thereafter, World War I broke out in 1918, and President Wilson, acting upon advice from the Department of Defense, decided to take control of the telephone and telegraph system as a wartime precaution. President Wilson appointed the Postmaster General to head the system. After one year, the control was returned to private hands, but in the process, the ICC gained regulatory authority. The good news for AT&T was that it demonstrated the importance of a national telephone system dedicated to universal service which played right into Vail's hands. [Ref. 4: p. 11]

H. THE COMMUNICATIONS ACT OF 1934 - REGULATION TAKES A FOOTHOLD

Vail never saw just how successful his strategy was because he died in 1920. By 1934, AT&T was solidly the leader in telephones with a market share of 80 percent. In support of Vail's concept of a regulated monopoly, Congress passed the Communications Act of 1934 that implicitly established the concept of universal service and acceptance of the telephone system as a natural monopoly. Universal service included three elements. First, the marginal utility for each subscriber was increased by the addition of other subscribers. In essence, the larger the network grew, the higher the value to each customer. Second, the telephone was viewed as a necessary service that would link the nation closer together. The comparison was often made to the contributions of the railroad. Third, the telephone was viewed as essential to each individual, not just the nation. This concept embraced the necessity to each person for daily activities and suggested the idea that the telephone could help individuals in need of assistance. [Ref. 5: p. 6]

Regulation brought with it several consequences. First, the subscriber could not own or attach equipment to the telephone since it was property of AT&T. Second, AT&T would not connect local independent TELCOs to their long lines, which created extreme difficulty for the TELCOs. Third, vertical integration by definition prevented non Western Electric manufacturers from supplying equipment. Fourth, telephone rates were based on cost plus a guaranteed ROR. Finally, regulation provided rate subsidies between local and long distance rates. All five of these consequences contributed to the plan of universal service.

Although the Communication Act of 1934 embraced the idea of regulation that provided market security for AT&T, it also proved to be a source of problems. As part of the act, the ICC transferred its regulatory authority to the newly established Federal Communications Commission (FCC). The FCC was granted broader powers than the ICC and was directed by President Roosevelt to investigate and report on AT&T and other companies involved in the telephone industry.

The commission gained momentum in 1936 under the FCC's Telephone Division Commissioner, Paul Walker. By 1938, Walker and AT&T had become adversaries, and the FCC was now threatening AT&T's monopoly position. On April 1, 1938, Walker released a draft report charging that AT&T had overcharged for services and equipment. The focus of the report centered on AT&T's manufacturing subsidiary, Western Electric. The report also pointed out that state regulatory agencies were failing to adequately control policy, and commented that government ownership should be considered. Finally, Walker sought broader, more sweeping control over AT&T with the power to approve or disapprove all contracts. The response to the draft was swift and negative. The members of the FCC clearly pointed out that they did not support his findings, and even national newspapers such as the Wall Street Journal labeled the report a travesty. The final outcome was the abolishment of the Telephone Division of the FCC and release of a watered down report in 1939. Regardless of the commission's final verdict, it did raise some serious questions regarding the structure of the Bell system. [Ref. 4: p. 12-13]

I. THE ANTITRUST SUITS BEGIN

The seeds planted by the Walker report finally came to fruition in 1949 when the government filed suit against AT&T, seeking to split Western Electric away. Once again, the focus of controversy was on the manufacturing arm where the government saw competition to be in the public's best interest. Needless to say, AT&T denied these charges. Ironically, a year later the Atomic Energy Commission (AEC) requested Western Electric to take over the government's Sandia Laboratory. The Department of Defense pursued this aspect in light of the rapid Russian nuclear buildup and wanted Bell Labs and Western Electric to accelerate their nuclear capability. AT&T used this as an argument against divesting Western Electric.

As time passed, AT&T and the government were not conceding anything. When the Korean War broke out in 1951, the Department of Defense again fought to dismiss the suit, arguing the need for a strong communication system was essential in time of national crisis. By this time, the Defense Department relied nearly 90 percent on AT&T for communications. Although the Truman administration agreed, the Justice Department would not concede. The issue remained deadlocked until shortly after the Eisenhower administration took office when the new Attorney General, Herbert Brownell, motioned for a compromise. Finally, on January 24, 1956, the Federal Court accepted a consent decree. The decree would not necessitate divesting Western Electric and would not alter AT&T's structure. However, AT&T was prevented from entering any business other than common carrier communications services, Western Electric was only to manufacture equipment for AT&T's use, and AT&T was required to license Bell patents to other TELCOs so they could manufacture their own equipment. In light of the circumstances, this was considered by many to be a major victory for AT&T. However, this victory was short lived. [Ref. 4: p. 14-15]

In November of 1956, the U.S. Court of Appeals ruled that customers had the right to use equipment with the telephone, other than those obtained through Western Electric. Up until this time, AT&T had successfully maintained a policy requiring exclusive use of Western Electric equipment in conjunction with AT&T services. The decision was based on a device called Hush-a-Phone, which was a cup-like attachment to the phone that provided the speaker privacy. AT&T and the FCC argued that such attachments would compromise transmission quality. The Hush-a-Phone ruling turned out to be an exception since the FCC and state public utility commissions (PUC) banned most other equipment. However, the key aspect of the Hush-a-Phone decision was that for the first time AT&T's end-to-end monopoly had been successfully challenged which set the stage for future suits. [Ref. 2: p. 31]

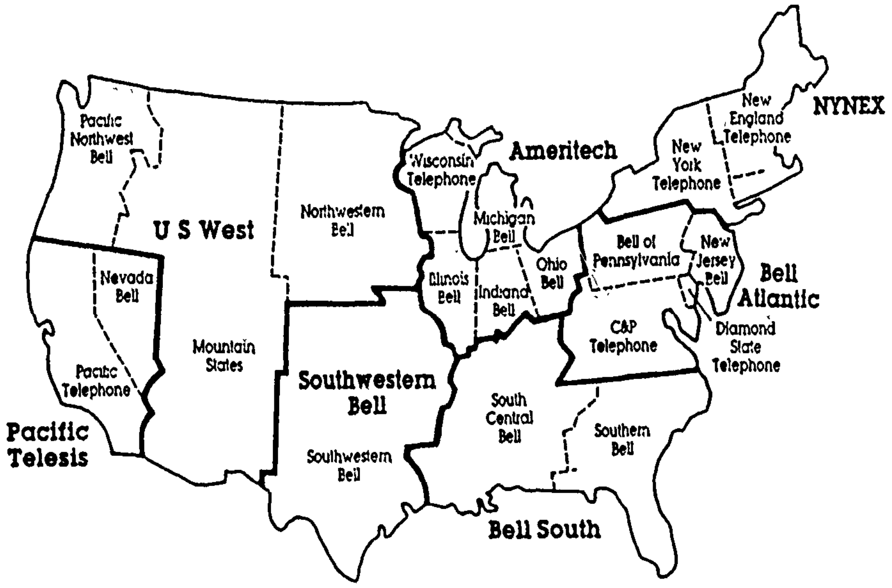
J. THE ROAD TO DIVESTITURE

For the next decade, AT&T enjoyed a rather stable environment free from lawsuits. That all changed in 1968 when two events affected it severely. First, a Presidential Task Force pointed out that divestiture of Western Electric was the best way to achieve diversity and competition in the telephone industry. Secondly, AT&T found itself back in court over an oil company's attachment to the telephone network. The device called the Carterphone was designed to link a mobile radio system into the telephone network. AT&T argued that it could not control the harmful effects to the network if devices such as this were allowed. However, in June of 1968, the FCC unanimously ruled in favor of allowing the Carterphone and other privately owned equipment connection to the network, provided that adequate protection for the network was ensured. This was a major blow to AT&T, who for 80 years had gone by a company policy requiring all equipment connected to the network to be Western Electric. [Ref. 2: p. 32]

During the 40s and 50s, AT&T thrived on technology including microwave, coaxial cables, direct dialing systems, the first transatlantic cable, and the transistor which its Bell Labs had invented. But now, technology was soon to be the next, and eventually fatal, threat concerning divestiture of subsidiaries of AT&T. The two technologies that affected it the most were computers and microwave, which ironically they had helped to develop. Back in 1963, a company called Microwave Communications, Incorporated (MCI) requested permission from the FCC to construct a long distance telephone system between Chicago and St. Louis. AT&T opposed the request and refused to provide local connections. MCI appealed to the FCC who ruled in their favor in 1969, but did not immediately rule on the issue of forcing AT&T into providing local connections. [Ref. 6: p. 4]

In 1971, the FCC gave approval to MCI to provide service and ordered AT&T to provide local connections. As a result, competing telephone companies turned to the antitrust laws more and more for protection. However, AT&T was not the only company being taken to court. One such instance involved International Telephone and Telegraph (ITT) and General Telephone and Electronics (GTE). By comparison, GTE represented only 8 percent of the telephone market, whereas AT&T represented 80 percent. Although much smaller, GTE had been using tactics similar to AT&T in that it would buy up independent TELCOs and vertically integrate the manufacturing capabilities. ITT, a non-integrated supplier of equipment, found itself with a shrinking customer base and appealed to the FCC on the grounds of the antitrust laws. The FCC ruled in favor of ITT and ordered GTE to competitively bid on all telephone equipment and to remain at arm's length from its manufacturing subsidiaries. Although AT&T was not affected by this ruling, its time was coming. [Ref. 2: p. 33-34]

Finally, on November 20, 1974, the Justice Department, concerned over the monopolistic powers of AT&T, got involved for good and filed a new anti-trust suit against AT&T. This suit was aimed at eliminating the monopoly and called for complete divestiture of Western Electric and some or all of the Bell Operating Companies (BOC) who handled the local service. For eight years and at a cost of $360 million to AT&T, the suit remained tied up in the U.S. District Court of Appeals in Washington, D.C., under Judge Harold F. Greene. At last, AT&T, realizing Judge Greene would not bend, offered to divest the BOCs if it could keep Bell Labs and Western Electric. In return, AT&T would be allowed to enter other lines of business once restricted by the 1956 consent decree. So, to the amazement of the nation, the Modified Final Judgment (MFJ) was agreed to in January 1982. The MFJ, referring to the 1956 judgment, contained the following provisions: by January 1, 1984, AT&T would divest all 22 of its local companies (BOCs), which were grouped into seven Regional BOCs (RBOCs), as depicted in Figure 1 on page 16; AT&T would end its virtual monopoly on long distance by ensuring the local BOCs would provide equal access to all competition; AT&T would be allowed to enter all telecommunications fields, including computers; and the BOCs were precluded from manufacturing and would continue their regional monopoly on local service. [Refs. 3: p. 336-337, 7: p. 4]



This figure shows the 22 BOCs and the seven RBOCs

Figure 1. Post-Divestiture Bell Operating Companies (BOCs) [Ref. 4: p. 1]

So, to the finish, AT&T essentially fought the same battles for nearly a century. AT&T is now organized into three divisions: AT&T Communications, which is the long lines division; AT&T Network Systems, which evolved out of Western Electric; and AT&T Information Systems, which diversified into the unregulated, high technology businesses. Incidentally, the Bell trademark was only allowed to be used by AT&T in its Bell Labs. All other uses of the Bell name and trademark went to the local BOCs. [Ref. 3: p. 336-337]

As for the competition, AT&T has continued to maintain roughly 70 percent of the market share of long distance, with the next closest competitor being MCI, followed by GTE Sprint. The "baby Bells," or BOCs, have done surprisingly well considering they were handed the most stagnant, low growth piece of the pie. However, in order to protect the "baby Bells," additional legislation was necessary to prevent bypass carriers (bypass carriers will be defined in the next chapter) from stripping away the profitable urban areas, leaving the "baby Bells" with the costly rural areas. [Refs. 8: p. 229,9: p. 70]

The BOCs have been aggressively diversifying into everything from retail store chains to publishing. This diversification has raised some concern over their priorities, but regulation and profits in the area of basic telephony have been strong enough and seem to have locked them onto the desired priorities. Finally, through the thick of the fight, over 1400 other local independent TELCOs have managed to survive. Most of these are contained in the rural areas of the nation where they started.

III. REGULATION

A. INTRODUCTION TO REGULATION

This chapter will provide a synopsis of regulation that is a critical force in the telephone industry and has shaped the structure of the network. Chapter 2 described the origins of regulation that were implanted by Theodore Vail and cemented in place by the Communications Act of 1934. However, regulation has endured radical changes caused by the Modified Final Judgment (MFJ) in 1982. As to be expected, the MFJ imposed the most stringent regulation on AT&T since the sole purpose of the MFJ was to eliminate the monopoly of AT&T in order to control the resultant entities through regulation. The purpose of regulation is to ensure the telephone companies provide the public with the quality of service desired at a fair price. Regulation is designed to achieve these goals by ensuring competition, restricting undesirable ventures, and providing incentives to channel efforts toward desirable behavior. Before discussing regulation any further, it is necessary to define who the regulators are and those who are affected by regulation.

B. THE REGULATORS

1. Federal Regulators

Regulators come in a wide variety, from the Federal Communications Commission (FCC) to the U.S. Congress. Regulatory bodies can be federal or state, and each has limitations on what it can regulate. However, Congress and the Justice Department seem to transcend these boundaries quite freely and control the situation. The first group of regulators to be discussed are the federal regulators.

The most prominent of the federal regulators is the FCC. It is the federal watchdog who are charged with enforcing the policy of the current presidential administration. The FCC was created by the Communications Act of 1934 and regulates all interstate and foreign telecommunication matters. It is responsible for the orderly development and operation of broadcast services and providing rapid, efficient nationwide and worldwide telephone services at reasonable rates. One function the FCC exclusively handles is radio frequency spectrum allocation for all private interests, including individual, business, and state and local government allocations. The FCC accomplishes this through its Private Radio Bureau. The FCC is an independent commission consisting of five members with one member who acts as the chairman and works directly for the president. Although there are over 2000 telephone companies in the U.S. today, the FCC regulates only 60. This is because the remaining companies are very small and regulation would achieve no positive benefits. [Ref. 8: p. 3-5]

Executive Order (EO) 12046 of 27 March 1978 transferred to the Department of Commerce the authority to formulate policies concerning telecommunications, promote efficient use and development of telecommunication services, and provide policy and management of the federal frequency spectrum. Furthermore, the Commerce Department develops long range plans for improved management of electromagnetic spectrum resources in conjunction with the FCC. In order to coordinate interagency telecommunications matters, the Secretary of Commerce created the Interdepartmental Radio Advisory Committee (IRAC) which is a working group consisting of representatives from interested agencies that act in an advisory capacity to the secretary. The EO also created the National Telecommunications and Information Administration (NTIA) to serve in three primary roles. First, the NTIA acts as the principal advisor to the executive branch on telecommunications matters relating to policy by providing technical support and advice. Secondly, the NTIA is responsible for coordinating allocation of that portion of the spectrum required by the federal government with the FCC. This includes the military spectrum. Third, it provides the executive branch with planning and research for the telecommunications needs of tomorrow. There has been a great deal of discussion among the various government agencies concerning the concept of universal service and the post-divestiture environment. It is widely recognized that the Communications Act of 1934 is outdated, and many industry experts feel that a new policy needs to be formulated. Presently, the NTIA is conducting a national study due to be completed in late 1990 that will recommend a national policy. Since the NTIA advises the administration on policy, and the FCC enforces this policy, there is a great deal of overlap. The head of NTIA is a political appointee who reports to the Assistant Secretary of Commerce for Communications and Information. [Refs. 10, 8: p. 4, 11 . p. 112-113]

The next two federal agencies are not directly in the regulation business, but have a major impact on regulation. Congress has been involved in telephone regulation since the beginning, especially after the passage of the Communications Act of 1934. Additionally, the MFJ provided increased incentive for it to get even more involved. Realizing after the MFJ took effect that the Communications Act of 1934 was getting obsolete, Congress passed HR 4102 by a voice vote in November 1983. The bill was created to sustain the concept of universal service, but unfortunately it lacked strength.

Further weakening the bill was the fact that the Senate never passed a similar bill. However, in the final analysis, Congress controls the FCC indirectly through passage of legislation and budgets the FCC must follow.

The Department of Justice is the last federal agency that is involved in telephone regulation. Up until the last two decades, the Justice Department has stayed out of the process of regulation. However, it was drawn into the fray several times by antitrust suits, but they were all settled out of court. Today, the Justice Department is heavily involved as a result of the MFJ and Judge Greene's decision to retain control of MFJ issues in the court system. Under the terms of the MFJ, the Justice Department is the approving agency for waivers requested by AT&T and the BOCs. [Refs. 8: p. 3-5, 12: p. 65]

2. State Regulators

The primary body concerning state regulators is the state public utility commissions (PUCs). Their span of regulation ranges from one to 168 different telephone companies they must regulate. The large variance is caused primarily by the number of telephone companies in respective states, and by the state PUC's decision whether they feel the company needs to be regulated. Some states do not regulate tiny telephone companies since the regulation enforced on the larger companies is designed to allow the tiny TELCOs to compete. State regulation began in 1885 when Indiana limited telephone rates to $3.00 per month. The trend for state regulation grew rapidly, and by 1922, 40 out of the 48 states had PUCs. Today, state PUCs regulate gas, electricity, and water, etc., in addition to telephones. Where the FCC is concerned with interstate regulations, the state PUCs are concerned with intrastate regulations. Also, there is a wide range of involvement by state PUCs in telephone regulation. For rural states with only a few companies to regulate, the job is relatively easy. However, the MFJ has created a significant workload for most state PUCs. Once overseeing a virtual monopoly, they now have to watch over several telephone companies that have different stakes. Lastly, the commissioner(s) of the state PUC is usually a political appointee(s).

As to be expected, trying to get 50 states all to agree on a single issue rarely occurs. Therefore, there are occasional disagreements between the PUCs and the FCC. For example, the PUCs have disagreed on every major FCC policy decision since the 1968 Carterphone decision. However, the FCC has what is called Federal Preemption, which stems from the Federal Supremacy Doctrine. If the state PUC still is not satisfied, the PUC can get the courts involved. The FCC decision must be appealed to the U.S. District Court of Appeals in Washington, D.C. (Judge Greene's court), up through the

Supreme Court. On the other hand, if the FCC doesn't like a state law, it can appeal to the state courts for a ruling, and by the appellate process, all the way up to the U.S. Supreme Court. In this way, telecommunication laws are made.[Ref. 8 : p. 1-8]

Last in the list of regulators are the local governments. In some states, the state constitution has provided for a "home rule" clause that allows local governments to set telephone rates for their city or municipality without intervention from the state PUC. However, this form of regulation is not too common, and even if available, most cities defer rate scheduling to the state PUC. [Ref. 8: p.3-5]

C. LOCAL ACCESS TRANSPORT AREA AND EXTENDED AREA SERVICE

When the MFJ took effect on 1 January 1984, so did the concept of a Local Access Transport Area (LATA). The actual term LATA was not in the original MFJ, but later was created in an opinion concerning the MFJ. It is similar to an exchange area. An exchange area is the smallest geographical division set up by the state PUC. The size of the exchange is determined by population density and can vary significantly in size. The key feature is that only one local telephone company can serve that exchange. LATAs are geographic areas established by a specific BOC conforming to conditions set forth by the MFJ. The first condition was that LATAs are roughly equivalent to Standard Metropolitan Statistical Areas (SMSA) and must get court approval to expand beyond these court established boundaries. Secondly, LATAs do not cross state boundaries as a general rule. Third, a LATA may contain more than one local exchange area. There are approximately 190 LATAs today. [Refs. 8: p.45-50,12: p. 62,101]

The reason for LATAs was to divide the assets of AT&T between the BOCs at divestiture. Since AT&T was prevented from providing intra-LATA service, this gave the BOCs exclusive rights to intra-LATA service. There were exceptions to this division of service when it was deemed necessary to allow BOCs to carry inter-LATA traffic for technical reasons or to carry the signal to a neighboring independent TELCO that has no other means to connect to the national network. Also, since large cities are often situated near state borders, there are exceptions to the state boundaries for inter-LATA calling. [Ref. 8: p. 45-50]

Confusion often exists over LATAs and rate areas. The rate area is another name for the free calling area. Within each free calling area there may be several local telephone exchanges which are represented by the first three digits of the telephone number. To gain an appreciation for this, some exchanges do not even cover an entire city, whereas some LATAs cover the entire state. Therefore, the free calling area is not the same as a LATA. If the local subscribers desire to have their rate area expanded so they can call customers in a foreign exchange (FX), they must bring the issue to the attention of the state PUC. This expanded area is called Extended Area Service (EAS). The MFJ as a general rule outlawed any inter-LATA EAS, but allowed a loophole for such requests. Once the state PUC determines an EAS request meets the requirements of the state's laws, the PUC requests the local telephone company to petition the Justice Department for approval. Once the Justice Department has approved the request, subscribers will be allowed to call into the FX free of charge. However, this added service will result in higher local telephone bills. [Ref. 8: p. 81-83]

Since LATAs were created as a method to split up AT&T, they only apply to the BOCs and to General Telephone Operating Companies (GTOCs). The reason GTOCs are included is simple. General Telephone and Electronics (GTE) offered to buy Southern Pacific Railway Corporation's telephone network called SPRINT in 1982, just after the MFJ decision. Until that time, GTE's telephone business consisted of only the GTOCs, which also manufactured their own equipment. Once GTE purchased SPRINT, it was able to provide long distance service and was totally vertically integrated, just like AT&T before divestiture. Why was a company allowed to structure itself after a monopoly that was being divested? The answer is due to the fact that GTE was much smaller, the GTOCs were more widely scattered, and GTE had less market power than AT&T. Another factor was that SPRINT represented only one percent of long distance service at the time of approval in 1982. However, the approval to purchase SPRINT required GTE to agree to conditions that included no cross subsidies between the GTOCs and SPRINT, equal access to the GTE network for other carriers, and other conditions similar to those placed on AT&T by the MFJ. [Ref. 8: p. 36-38]

Since the BOCs and GTOCs use LATAs and the small independent TELCOs do not, there are seven different situations for regulation. The first two situations are inter-LATA/interstate and inter-LATA/intrastate. Both situations require an Interexchange Carrier (IXC) to connect the two companies. The next two situations are intra-LATA/interstate and intra-LATA/intrastate. Both situations normally require a Local Exchange Carrier (LEC) to connect. The third set of situations involves service from a non-LATA area to a LATA or from a LATA to a non-LATA area. Because a BOC or GTOC is providing the local exchange service, an IXC must provide the connection. Finally, the last situation is non-LATA to non-LATA where no restrictions apply since this would be a call between two unregulated independent TELCOs. IXCs and LECs will be explained next.

D. THE REGULATED

The regulatory environment is broken down into three broad regulation categories: Interexchange Carriers (IXCs), Local Exchange Carriers (LECs), and Customer Premise Equipment (CPE). An IXC carrier is one that handles long distance communications. Examples of IXCs are AT&T, MCI, and SPRINT. LECs are the providers of local service within a LATA or local exchange. Examples of LECs are the BOCs, GTOCs, or other local independent TELCOs. CPE companies provide the terminal equipment such as the telephone terminal, Private Branch Exchange (PBX), or Key Telephone Set (KTS) to end users. Examples of CPE companies are AT&T, Rolm, and Mitel. Figure 2 shows how the IXC, LEC, and CPE concept looks graphically. As can be seen, an IXC carrier is required to connect the two LATAs. The Central Office (CO) in each LATA is the central switching facility that switches inputs from the customer into the network. The Point of Presence (POP) is where the IXC carrier has placed his switch to connect calls from that LATA to the long distance network. Each IXC company will have a POP in each LATA. [Ref. 12: p. 62 and 100]

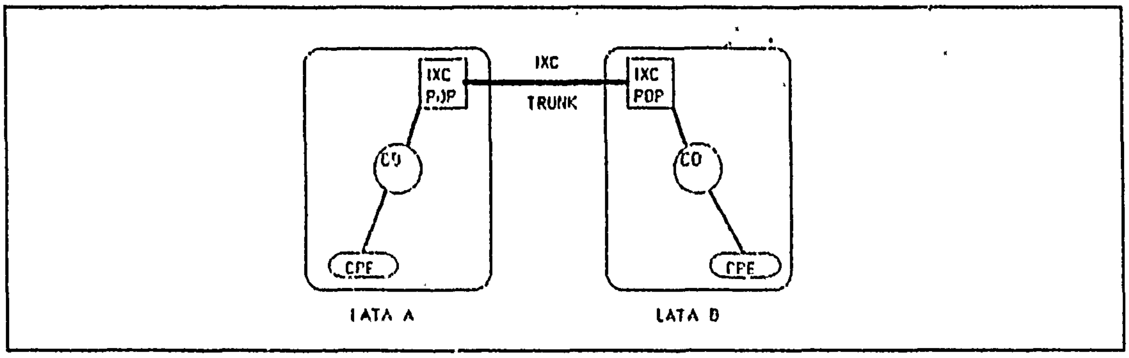


Figure 2. Relationship Between LATAs, IXCs, LECs, and CPE

However, within the broad categories of IXCs and LECs, a company can get another classification. The most prevelant classification is "common carrier." The definition of a common carrier is a company open to public hire to provide intrastate, interstate or foreign communications by electrical means [Ref. 8: p. 10]. In other words, the BOCs, GTOCs, AT&T, MCI, and SPRINT are all common carriers. As a general rule, regulation for intrastate telephone services are accomplished through state PUCs, whereas interstate matters are regulated by the FCC. Regulation is a result of a common carrier's particular situation. The three main reasons to regulate common carriers are: they are a monopoly; they use a scarce frequency spectrum resource; or as in the case of AT&T, their size dictates regulation due to their enormous market power. When a company's size dictates regulation, it is called a dominant carrier. [Refs. 8: p. 10, 12: p. 100-110]

One term often used is Other Common Carriers (OCCs). This term evolved during the legal battles leading up to the MFJ when AT&T was the defendant common carrier and the remaining common carriers were the OCCs. Until the MFJ, all carriers were regulated alike. However, the MFJ recognized the dominant carrier aspect of AT&T, and now AT&T is literally being regulated by the FCC in a class by itself. Additionally, most state PUCs regulate AT&T through separate actions, but their trend is now shifting back towards deregulation of AT&T. [Ref. 8: p. 10-11]

The other types of regulated services are cellular radio common carriers, bypass carriers, satellite carriers, and resellers. The cellular radio market is regulated by limiting only two franchises to market their products. One of these franchises must be awarded to the local BOC, GTOC, or independent TELCO. This allows local carriers to provide the landline portion of the network which also must be provided on an equal access basis to their competition. The mandate that only two franchises be allowed into an area has created many joint ventures. Resellers are simply companies that lease facilities from a larger common carrier and package the product differently. An example of this could be leasing large capacity lines and then subdividing the capacity or bandwidth to sell in smaller blocks. Resellers that add an extra service, such as some software services, are called Value Added Carriers (VACs). [Ref. 9: p. 70-71]

The two remaining carriers are specialized common carriers and consist of bypass carriers and satellite carriers. Bypass carriers usually go after the high density traffic areas around or between large metropolitan areas. Basically, the motivation for the bypass carrier is to provide lower rates by bypassing the LEC CO, as shown in Figure 3 on page 25. The bypass carrier can also bypass the IXC carrier and go directly to the destination LATA's CO or CPE. The bypass carrier has been regulated to prevent stripping away the high profit routes from the BOCs, GTOCs, and independent TELCOs, leaving them with only the high cost routes. The bypass regulation has led to the concept of access charges which will be discussed later. Like the bypass carrier, satellite carriers perform a bypass function. Again, satellite carriers can bypass both the LEC CO and the IXC POP by purchasing or leasing satellite antennas and placing them on the subscribers' property. This way the CPE can be wired directly to the antenna which uses a satellite to relay the traffic to another satellite antenna. For small businesses, this may be too costly, so their solution is to bypass using a satellite link between the LEC COs. The complete satellite carrier bypass situation is depicted in Figure 4 on page 25.

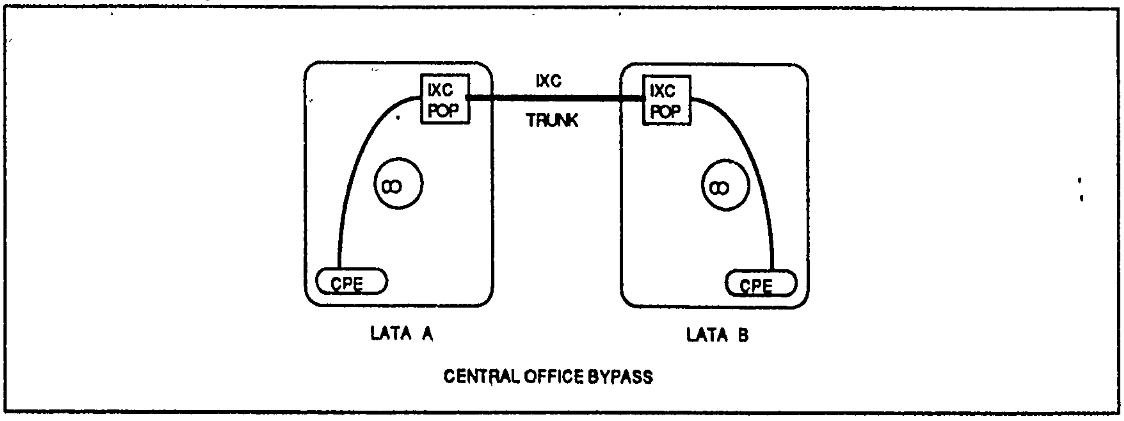


Figure 3. Bypass Carrier Bypassing the Central Office

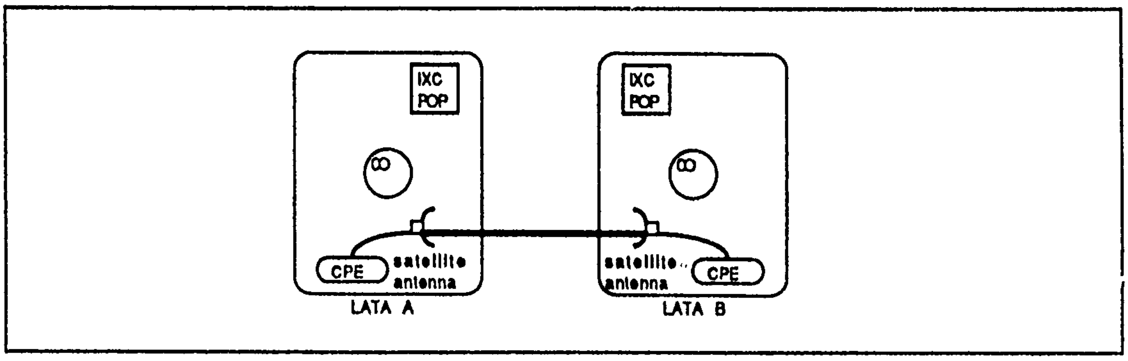


Figure 4. Satellite Carrier Complete Bypass

E. THE REGULATION

1. Tariffs

Usually the outcome of regulation results in a tariff. Tariffs are essentially contracts between the regulatory agency and the company providing the service. Tariffs specify rates, charges, standards, or any regulation deemed appropriate. The bodies that approve tariffs are the FCC for interstate service and the state PUCs for intrastate service.

2. Rate of Return

The Rate of Return (ROR) is the most common way to control a carrier's profits. The MFJ prompted a three phase plan to control ROR. Phase I occurred immediately after the MFJ took effect and constituted full regulation. Presently, we are in

Phase II which is the transitional competitive stage that established rate bands for RORs. This meant the carrier was limited by a ROR ceiling and floor. For example, the California PUC specifies a minimum ROR around 8.5% and allows the company to retain all earnings up to 16%. Any earnings over 16% are shared with the rate payers by crediting their accounts for the subsequent period. The FCC conducts rate reviews and modifies the pricing structure to maintain ROR within the band. These rate bands provide an incentive for the carrier to provide the services at a reduced cost. The next step in Phase II regulation is price level regulation and will be discussed in the next section. Phase III will be the fully competitive stage where the company essentially regulates itself. [Ref. 13]

3. Price Level Regulation

Price level regulation is similar to ROR regulation in that it generally provides a target range for the company to set its prices. The basic concept of price level regulation is to allow the company some flexibility in its pricing, thus moving one step closer toward deregulation of the market. Price level regulation can be in the form of price ceilings or price floors. Price ceilings are computed based on costs submitted to the PUC with a percentage added for profit. Price floors are used in cases where a stronger competitor could price its services below cost in an attempt to drive a weaker company out of business. The key to price level regulation is competition. Services that have significant competition are using price level regulation as opposed to ROR. Some examples of these services are the call forwarding and call waiting features. As soon as competition gains a foothold in a particular market, price level regulation will be introduced. Already, several states have introduced price level regulation in the inter-LATA toll call market. The next step after price level regulation is phase III discussed above which will occur in a fully competitive market. [Ref. 14]

4. Minimum Telephone Service Standards

In order to give an idea of what aspects of telephone service are being regulated, the following partial listing represents minimum telephone service standards. The areas include:

- General provisions
- Definitions
- Records and reports
- Tariff contents
- Applications for upgrade of service

- Customer complaints
- Directories
- Billing
- Adequacy of service
- Quality of service
- Directory assistance
- Public telephone service
- Plant design
- Courtesy
- Maintenance
- Interruption of service
- Transmission standards
- Public information
- Emergency operation

5. Costs

In the years before the MFJ, local rates were heavily subsidized by long distance calls. In the post-MFJ years, local call subsidies have disappeared as the telephone bill today is allocated differently. Today, a telephone bill consists of three parts: a fixed monthly charge, a usage charge, and an access charge. The fixed charge represents the payment to the local telephone company or LEC. The rates for the fixed charge are set by the state PUC. The usage charge is based on connection time, duration of connection, and the airline mileage band. These charges apply only to long distance calls and are set by state PUCs for intrastate and by the FCC for interstate long distance. The last charge is the access charge, which came about as a result of the bypass threat, and has effectively supplanted the cross subsidization that once was effective in offsetting the local costs. These three costs are for switched services over the Public Telephone Network (PTN). If the customer is leasing a dedicated line, the costs will include: a fixed monthly charge; a mileage charge; and an access charge. The difference is that the leased cost is not usage dependent, or as they say, is a non-traffic sensitive (NTS) cost.

If the user can qualify for bulk rate service, there may be an opportunity to reduce costs. The bulk rate service over the PTN is called Wide Area Telephone Service (WATS). There are two types of WATS: INWATS and OUTWATS. INWATS is another name for an 800 service, where callers can call a destination free of charge. OUTWATS is the opposite, where a company can call out at a reduced cost. The discount is based on the volume of traffic in or out and the Service Area (SA) the user is calling to or from. Service areas range from SA1 through SA6. SA1 is defined as all adjacent states where SA5 is all contiguous states, plus Puerto Rico and the Virgin Islands. SA6 is all 50 states.

IV. BASIC TELEPHONE TECHNOLOGY

A. NETWORK OVERVIEW

1. Design Considerations

The telephone network is the largest communications system in the world. As discussed in Chapter two, development of the telephone system has been an evolutionary process that has resulted in the structure of the network as we know it. Today, there are several important aspects that must be considered in network design. Primarily these aspects are based on existing facilities and traffic statistics. The considerations applicable to existing facilities include the number of terminals, speed of the facilities, and physical location of the equipment. Traffic statistics include average number of terminals in use, speed of equipment, average number of calls, length of calls, and peak utilization periods. Once these design considerations have been determined, the topology of the network is determined. Aspects of the network topology that need to be analyzed are physical considerations, cost considerations, and network requirements. Physical considerations are concerned with terrain features and practicalities involved with certain equipment. Network requirements include growth, redundancy, and capacity. The most critical decision is the design of the backbone, or primary network. [Ref. 15: p. 103]

Once the network is designed, an analysis must be conducted to include a congestion study. Telephone companies design networks and are regulated to provide 99.998% chance of success in connecting the call under normal circumstances. If too few trunks (main cables) are provided, congestion may cause blocking which results in a lost call. This means no connection can be made and the call is "lost" indicated by the user getting a busy signal. This requires the user to redial. To analyze congestion potential, it is assumed the calls are made stochastically and a poisson model is used to determine the required number of trunks to avoid congestion, given a call completion probability of nearly 100%. In order to determine the requirements, a unit of measurement called an erlang is used to reflect traffic intensity. The erlang measures the number of circuits in use at a specified time. Therefore, the number of erlangs that pass in one hour is given by $A = \lambda t$, where A is the number of erlangs, t is the average time of a call in hours, and λ is the total number of calls in one hour. For example, if A is .3, this means that the circuit is 30% utilized. The erlang is an international standard, whereas, hundred call seconds (CCS) is an American standard. Since there are 3600 seconds in one hour, this equates to 36 CCS. [Ref. 16: p. 1-15]

2. Basic Architecture

The basic architecture is shown in Figure 5 on page 31 where two telephones are depicted connected via the long distance network. As a general rule, twisted pair is installed between each CPE and the serving CO. Trunks can utilize many different mediums and will be discussed in a later section. The lines between the CPE and CO are referred to as the local loop and are located in the LEC, whereas the IXC is essentially between the two COs. The architecture design provides for reliability through redundancy and diversity. Redundancy is provided by installing standby equipment or lines in the event of a failure or unforeseen congestion. The type of diversity that is used depends upon what kind of transmission is involved. Types of diversity are space, frequency, angle of arrival, polarization, time, multipath, and method diversity. Space diversity concerns the physical location of the transmitters and antennas. In the case of antennas, space diversity can be achieved by horizontally separating them or vertically stacking them which is often used in microwave transmission. Frequency diversity is used to avoid atmospheric conditions that affect the transmission of different frequencies and is used in satellite and microwave systems. Angle of arrival diversity is used to avoid degrading effects caused by "looking" into the sun or to avoid low antenna angles and is used mostly in satellite transmissions. Polarization diversity is used to optimize the effects of polarization in transmission. Time diversity is when the same information is repeated or transmitted twice over the same network. Multipath diversity involves the simultaneous transmission of identical information over different paths. This type of diversity is used in microwave and satellite transmission since rain will degrade their signals dramatically. By separating the paths, the chance of a rain cell blocking both paths is minimal. Finally, method diversity involves transmission of the same signal via two separate means, for example, coaxial and microwave. [Ref. 17: p. 208 and 452]

3. Telephone Basics

There are many types of telephones, but all operate on the same principles established over 100 years ago. In Figure 6 on page 31, the user speaks into the transmitter that uses the changes in air pressure to vibrate the diaphragm. This diaphragm uses the vibration to expand or contract carbon granules located in the transmitter housing. This expansion and contraction changes the resistance proportional to the user's voice and is converted into an electric signal varying in strength. The power to transmit this signal comes from a DC talk battery located at the CO. The signal is then

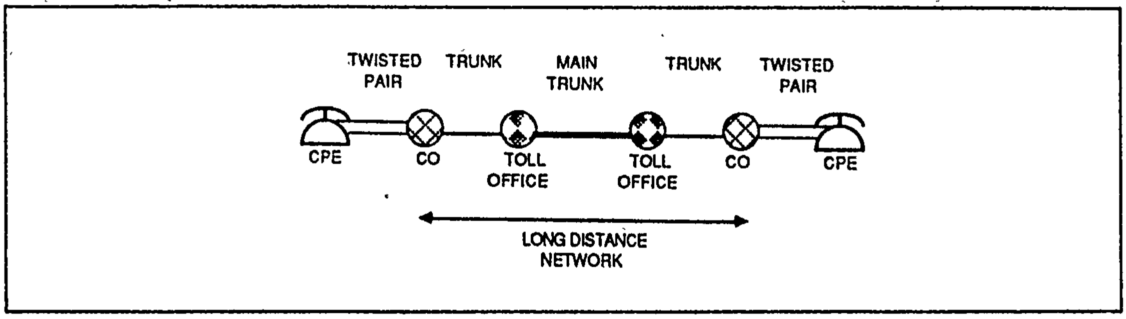


Figure 5. Simplified Basic Telephone Architecture

received by the receiver, which has an electromagnet attached to a metal diaphragm. The changes in the transmitted signal's strength causes the diaphragm to vibrate, replicating the speaker's voice. [Ref. 18: p. 8-9]

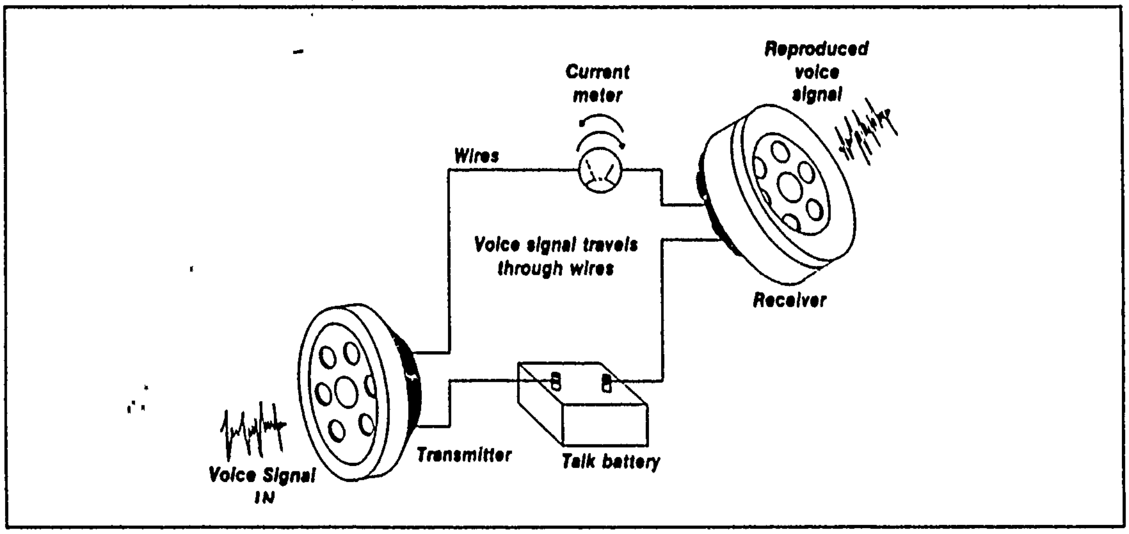


Figure 6. Simple Schematic of Telephone Handset [Ref. 18: p. 7]

Since the distance between COs will vary, a device must be made to compensate for signal loss. The concern is actually the lack of signal loss, since the CO provides enough voltage to operate even the most remote telephones. The problem is to control the volume for short distance calls. This problem is eliminated by a device called a varistor, or variable resistor, which is located in the telephone handset. Varistors also reduce the sound levels of dialing pulses or tones so the user can comfortably use the telephone.

Inside the premises, the wiring consists of four wires, usually 24 or 26 guage. The most important two are the green (tip) and the red (ring) wires, which are usually the only two that need to be connected. The determination to use two or four wires is determined by the manufacturers of the telephone terminal equipment. The "tip" is a term that was derived from the tip of a plug that operators once used to plug into a switchboard jack. Together the tip and ring circuits provide AC and DC voltage to operate the ringer (AC) and transport the voice signal (DC). From the telephone connection box, the lines proceed to a Standard Network Interface (SNI) or Demarcation Point Annex (DPA), which is the point where the telephone company's wiring is connected to the subscriber's. The DPA is usually a modular jack located near the power meter and has a lightning arrestor as part of the unit. The number of connections to a single DPA is limited by the Ringer Equivalence Number (REN) which is a number located on the bottom of the telephone. As a general rule, five telephones, or a total of five RENs, is the maximum that can be powered with a single drop (line from telephone pole to house or business). However, some older telephones require an REN greater than one per telephone, and five may be too many. One way to beat this is to disconnect the red (ringer) wire on adjacent telephones, as this only prevents that telephone from ringing. [Ref. 7: p. 49-60]

4. Two Wire and Four Wire Lines

Two wire lines (one pair) are installed between the CPE and the CO. The basic requirement for a telephone conversation requires two way transmission of information which means both signals must travel via the same wire pair for half duplex operation. Half duplex operation is defined in North America as transmission in two directions, but only one direction at a time. In a two wire system, if both users talk at the same time their conversation will be superimposed on the same wire and will be heard, but not understood. To achieve full duplex operation, a four wire system is used so that each signal has a separate path. Full duplex is defined as a system that can provide transmission in both directions simultaneously. Four wire systems are used for virtually all links in the telephone system and usually terminate at the CO. This means that one pair is used for transmitting and one pair is used for receiving. The conversion from two wire to four wire is conducted by a terminating set, usually just called a term set. One exception to the two wire CPE to CO is found in some military Autovon lines that are connected by four wire links. [Ref. 17: p. 42]

B. TRANSMISSION

1. Analog Versus Digital

The costs associated with transmission are approximately 50% of the total telephone system costs. In the telephone network, transmission is achieved by passing analog or digital data via an analog or digital signal through various media. Data is defined as the elements that convey the information, and signals are the electromagnetic encoding of the data. Signaling is the act of propagating the signal through modulation. An analog signal is defined as a continuously varying sinusoidal electromagnetic wave propagated through a variety of media, including twisted pair, coaxial, fiber optic cable, and atmospheric propagation. Digital signals are the sequence of binary pulses propagated by a metallic wire medium. Note that digital signals can only be propagated over wire. Therefore, a large portion of this section will be devoted to the transmission of analog signals that carry analog or digital data. [Ref. 19: p. 67-93]

In the case of analog transmission using an analog signal, the transmission is sent regardless of data type, however, digital data will require a modem. Either way, as the signal propagates along the medium, it begins to attenuate, or weaken. In order to boost the signal to continue propagation, the signal must be amplified. The problem is this process also amplifies any noise that is present making it hard to strip away or filter out the noise at the receiver. After a signal is passed over a long distance, the signal may become totally unreadable due to noise buildup. If the data is digital, noise that exceeds the threshold between binary values can cause catastrophic damage. This is because changing just one bit can change the entire meaning of the digital signal. Fortunately, voice can absorb a lot of noise and still be understood. Analog transmission using digital signals is not used.

The other type of transmission is digital, which can transmit either analog or digital signals. For the digital signal, attenuation occurs quickly and repeaters must be used to ensure that data integrity is maintained. To do this, the signal is received by the repeater, and the repeater recovers the binary code and retransmits the signal. The same method is used for an analog signal carrying digital data.

Today's technology is quickly changing to the digital transmission method. The reasons for this are: digital technology uses Very Large Scale Integration (VLSI) which means miniaturized equipment with dramatically reduced equipment and maintenance costs; repeaters used with digital technology do not amplify noise and are relatively inexpensive compared to analog amplifiers; digital transmissions are easier to multiplex using time division multiplexing (TDM) versus the more costly analog style frequency division multiplexing (FDM); encryption is easier using digital data; and both analog and digital data can be treated the same using digital signaling.

Often confusing is the reference to channel capacity or bandwidth. In the telephone system, capacity will usually refer to digital means and bandwidth refers to analog means. The problem then is how to convert back and forth when dealing in a mixed environment or transmitting mixed voice and data. The answer lies in a simple formula based on the Nyquist rate. The Nyquist rate is twice the highest frequency contained in the signal. Therefore, the formula is $C = 2W\log_2 M$, where C is the channel capacity in bits per second (bps), W is the bandwidth required, and M is the number of discrete signal levels that correspond to the type of encoding scheme. [Ref. 19: p. 44]

2. Data and Signals

Since it is possible to have digital or analog data in conjunction with digital or analog signals, this creates four different possibilities. First, digital data and digital signals are a combination used because digital equipment is cheaper and easier to maintain as a rule. The digital signal is a series of discrete voltage pulses, each pulse being a signal element. Each signal element represents a binary digit. There are two types of signaling that can be used, unipolar and bipolar. Unipolar pulses are all either positive or negative, whereas bipolar pulses are represented by both positive and negative. Another factor to consider is the digital encoding scheme, which includes Non Return to Zero (NRZ), Biphase L/M/S, Differential Manchester, and Return to Zero (RZ). Performance for digital systems is determined by data rates, signal to noise ratio (S/N), and bandwidth. The higher the data rate, the higher the Bit Error Rate (BER). Also, the lower the S/N ratio, the higher the BER. Increasing bandwidth increases the data rate. Digital data with digital signals are only sent over metallic wire mediums. [Ref. 19: p. 79-81]

Digital data can also be used with analog signals. This is one of the most common methods and is used every day by persons using a home computer hooked up to the telephone network via a modem. The modem takes the digital data from the computer and converts it into an analog signal so that the signal can be received, switched, and transmitted by the telephone network. The key aspect is that this type of transmission must be used for optical fiber and the unguided media. [Ref. 19: p. 75-81]

Next is the method using analog data and digital signals. The process to do this is called digitization, where the analog data is converted into digital data. Once the data is digitized, it can then be transmitted directly using NRZ, encoded using another scheme, or converted into an analog signal. Devices that convert analog data into digital signals are called coder-decoders (CODECs). Techniques used by CODECs are Pulse Code Modulation (PCM), Differential PCM (DPCM), Adaptive DPCM (ADPCM), Continuously Varying Slope Delta modulation (CVSD), and Vector Quantizing Code (VQC). PCM is the basic technique used in achieving most of these modulation schemes. PCM is a two step process where the technique of Pulse Amplitude Modulation (PAM) is employed to sample the analog data at a rate of 8,000 samples per second. Then each sample is digitally encoded to represent the strength of the sample using eight bits, thus requiring a data rate of 64Kbps to support PCM. DPCM recognizes the fact that voice signals change slowly and uses four bits vice eight bits making the required channel capacity only 32Kbps. By using VQC, the data rate can be reduced to as little as eight Kbps. However, all these techniques that reduce capacity requirements result in a tradeoff of increased noise. Therefore, the highest quality sound will be produced by straight PCM. [Ref. 20]

The reason the sampling rate is generally at least 8,000 samples per second comes from Shannon's sampling theorem which gives the Nyquist rate. Simply stated, the sampling frequency is equal to twice the highest signal frequency. Since telephones use bandwidths of 4,000 Hz, twice this yields a minimum sampling rate of 8,000 samples per second. If fewer samples are used, a problem known as aliasing can result when the signal is multiplied during the transmission process. An alias is an unwanted byproduct of the multiplication process that will overlap with the desired signal and destroy or distort the signal. Using a sampling rate of exactly twice the highest frequency to be passed assumes that frequencies above this range can be filtered out. Unfortunately, filters cannot be built with a precise enough pass band to eliminate all these frequencies. In other words, the filter's skirts are not sharp enough. Therefore, a guard band is used. In telephone transmissions, the information is actually contained in the range from 300 to 3400 Hz which is fit inside a 0 to 4000 Hz band that provides a guard band on each side. [Ref. 21: p. 266-267]

The final method involves analog data and analog signals. Voice in its natural form is analog data. The effective voice range is considered to be roughly 0-3400 Hz which is not high enough to transmit over most mediums. Therefore, an analog carrier is added to make the analog data into an analog signal. This is a very common method and was used almost exclusively until digital methods became popular. Modulation techniques used for this method are Amplitude Modulation (AM), Phase Modulation (PM), and Frequency Modulation (FM). [Ref. 19: p. 32-90]

3. Analog Transmissions

a. Amplitude Modulation

Amplitude Modulation (AM) is defined as the process where the instantaneous amplitude of a higher frequency carrier is varied in accordance with an information signal [Ref. 21: p. 123]. This type of modulation includes four basic types within the category of AM: conventional AM; Double Sideband (DSB); Single Sideband (SSB); and vestigial sideband. For simplicity, this section will focus on DSB AM, since all types of AM are basically alike. First of all, some definitions are needed and are listed below. [Ref. 21: p. 123-127]

- $x(t)$ = modulating signal at transmitter
- $y(t)$ = modulated signal at transmitter
- $y_r(t)$ = modulated signal at receiver
- $x_d(t)$ = detected signal at receiver
- $y_1(t)$ = output of receiver multiplier
- f_c = carrier frequency
- ω_c = carrier frequency in radians
- f = input frequency

To generate the DSB signal, the input signal is multiplied by a sinusoidal carrier with frequency f_c. The multiplication process is conducted in a device called a balanced modulator. The results of multiplying the input signal $x(t)$ with the sinusoidal carrier yield: $y(t) = x(t)\cos\omega_c t$. Since $x(t)$ is also a sinusoidal function, the resultant of multiplying two sinusoids is given by the trigonometric identity: $\sin(A)\sin(B) = 1/2[\cos(A-B) + \cos(A+B)]$. There are two similar identities used when the input and carrier sinusoids are cosine functions, or mixed. By conducting Fourier analysis of the frequency spectrum, it can be seen that there has been a translation of frequency shifting either side of the carrier frequency. The resultant frequencies are f_c-f and f_c+f, each of which is only one half the original amplitude. It can be seen how this resembles the form of the trigonometric identity. From this aspect, it is called double sideband. SSB is where one of the sidebands is filtered out and left with either the upper or lower sideband. The bandwidth required to transmit a DSB AM signal is given by the formula, $B_t = 2B$, where B is the bandwidth of the original input signal. If SSB is employed , then $B_t = B$, so the required bandwidth is only as much as the signal itself. This is an important advantage. The process to recover the signal at the receiver is similar to the transmit operation. The received signal $y_r(t)$ is multiplied by a sinusoidal at the carrier frequency resulting in the output: $y_1(t) = y_r(t)\cos\omega_c t$. Passing this product through a low pass filter results in $x_d(t)$ which is directly proportional to the transmitted signal. [Ref. 21: p. 127-146]

Conventional AM uses a similar technique, but this time puts a summing circuit in front of the balanced modulator. The result is: $y(t) = A[1+mx(t)]\cos\omega_c t$, where A is a constant less than one, and m is the modulation factor that is input into the summing circuit with x(t). The range of m is from zero to one, which represents zero modulation to 100% modulation. Again a similar process is used to recover the signal at the receiver. [Ref. 21: p. 146-150]

b. Angle Modulation

Angle modulation differs from AM in that the amplitude of the composite signal is constant and the information is conveyed by changing the angle of the carrier function. There are two types of angle modulation: frequency modulation (FM), and phase modulation (PM). The biggest advantage of angle modulation over AM is that it is easier to eliminate noise. Since the angle modulated signal has constant amplitude and noise tends to build up on the top of a signal during transmission, it is a simple matter to "clip" the top of the signal, thus eliminating the noise. Before proceeding further, it is necessary to define terms associated with angle modulation. [Ref. 21: p. 181-183]

- y(t) = output signal
- ω_c = radian frequency of the carrier
- $\phi_i(t)$ = instantaneous signal phase angle
- $\Phi_i(t)$ = total instantaneous signal phase angle = $\omega_c t + \phi_i(t)$
- $\omega_i(t)$ = instantaneous signal radian frequency
- $\Omega_i(t)$ = total instantaneous radian frequency = $\omega_c + \omega_i(t)$
- x(t) = normalized modulating signal = $\cos\omega_m t$
- $\Delta\phi$ = maximum phase deviation in radians
- $\Delta\omega$ = maximum radian frequency deviation
- $\Delta f = \Delta\omega/2\pi$
- f_m = modulating frequency (single tone)

For both FM and PM, the basic equation starts with $y(t) = A\cos[\omega_c t + \phi_i(t)]$. As mentioned earlier, the information is passed by altering the instantaneous phase angle, $\phi_i(t)$. Using PM, $\phi_i(t) = \Delta\phi x(t)$. Substituting this into the basic equation yields: $y(t) = A\cos[\omega_c t + \Delta\phi x(t)]$, where $x(t) = \cos\omega_m t$. Since ω_c and $\Delta\phi$ are constants, only the input signal x(t) will vary causing the angle to change, thus conveying the information. In FM, $\phi_i(t) = \Delta\omega\int_0^t x(t)dt$, where $x(t) = \cos\omega_m t$. Therefore, $y(t) = A\cos[\omega_c t + \Delta\omega\int_0^t x(t)dt]$, where ω_c and $\Delta\omega$ are constants. Again, the only variable is the input signal which this time is integrated with respect to time before being applied. The bandwidth of an angle modulated signal is computed by using Carson's rule. Since angle modulation contains a term in the form of a cosine function, the frequency range will be much wider than AM. Carson's rule states that $B_t = 2(1 + \beta)\beta$, where B is the input signal bandwidth and β is the modulation index, which equals $\Delta f/f_m$. [Ref. 21: p. 183-198]

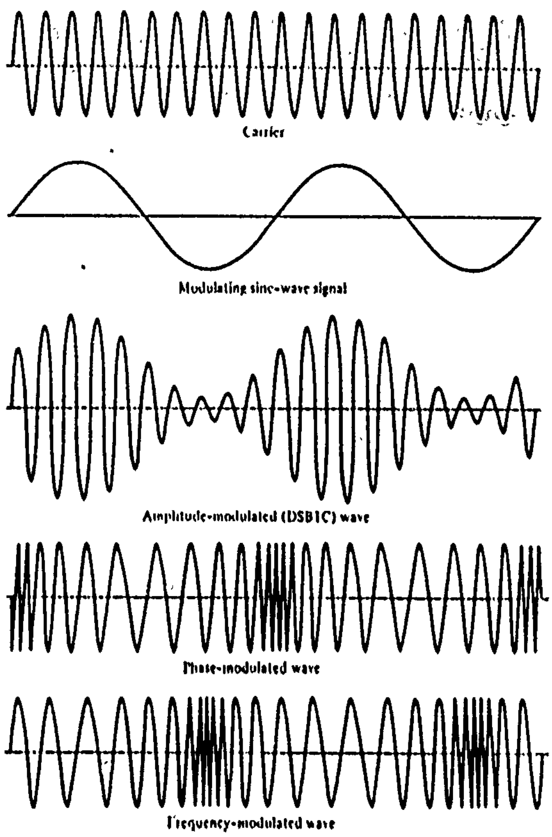
To better conceptualize the differences in the various analog modulation techniques, Figure 7 on page 39 is presented. The top line is the carrier frequency, which is chosen sufficiently beyond the human hearing range. As depicted, the carrier frequency does not change. The information is conveyed through a modulating sinusoidal signal that is either amplitude modulated, phase modulated , or frequency modulated. Notice that the amplitude modulated signal is the only one that changes amplitude and that frequency modulated and phase modulated signals look alike. Only by careful analysis, which requires comparing the carrier phase and frequency to the input signal phase and frequency and applying the basic formulas, can the difference be detected.

The reception of PM and FM signals is achieved by receivers called discriminators and detectors. Two of the more common types are superheterodyne and phase locked loop (PLL). A superheterodyne receiver is one that uses an intermediate frequency (IF) amplifier and filter to narrow the tuning band, eliminating the need for a highly selective variable center frequency filter that is difficult to build. Therefore, greater selectivity can be achieved by using the fixed frequency filter, designing it to have sharp cut off characteristics and a flat frequency response. PLL receivers are popular because they feature a feedback loop that enables the receiver to replicate the transmitter voltage more accurately which makes for a better quality signal reproduction. [Refs. 19: p. 191-194, 21: p. 181-248, 22: p. 11.1-11.11]

4. Digital Transmission

a. Introduction

Digital modulation or transmission is accomplished by sending a train of pulses representing the information signal. The pulses are of equal magnitude and duration and are encoded to represent bits. Digital transmissions are a good example of how the analog and digital worlds cannot be totally separated. The only practical



Top figure depicts the unmodulated carrier that is modulated by the input signal below it. The last three figures show AM, PM,and FM.

Figure 7. Analog Transmission Techniques [Ref. 19: p. 93]

method to send digital data by electromagnetic means is by using analog signals. [Ref. 21: p. 323]

b. Amplitude Shift Keying

In Amplitude Shift Keying (ASK), the two binary values are represented by changing the amplitude of the carrier frequency in response to the input function. The two most commonly used amplitudes are zero, represented by a binary 0, and some constant amplitude A, represented by a binary 1. For this reason, ASK is often called On-Off Keying (OOK) and is achieved by simply gating the carrier frequency. Using an analog signal of the form, $y(t) = A\cos\omega_c t$ representing the binary 1, and $y(t) = 0$ representing the binary 0, shows that ASK is achieved using a technique similar to amplitude modulation, but the difference is that the amplitude is not modulated. ASK is susceptible to sudden gain changes from noise and is only useful in lower data rate applications over voice grade telephone lines. However, ASK is used to transmit digital data over fiber optic cables. By analyzing Figure 8 on page 41, it is easy to see the corresponding on-off keying used in ASK. [Refs. 19: p. 75, 21: p.325]

c. Frequency Shift Keying

The process of Frequency Shift Keying (FSK) involves the transmission of two distinct frequencies. Usually these frequencies are offset from the carrier frequency in opposite directions by equal amounts. The lower of the frequencies can then represent the binary 0, and the higher frequency can represent the binary 1. This requires the transmitter to switch between these two frequencies at a high rate. The equations for FSK look like this: $y(t) = A\cos\omega_1 t$, representing the binary 0; and $y(t) = A\cos\omega_2 t$, representing the binary 1. Figure 8 on page 41 shows this relationship. FSK is less prone to error than ASK and is often used in radio transmission and coaxial cable applications. [Refs. 21: p. 324-325, 19: p. 76]

d. Phase Shift Keying

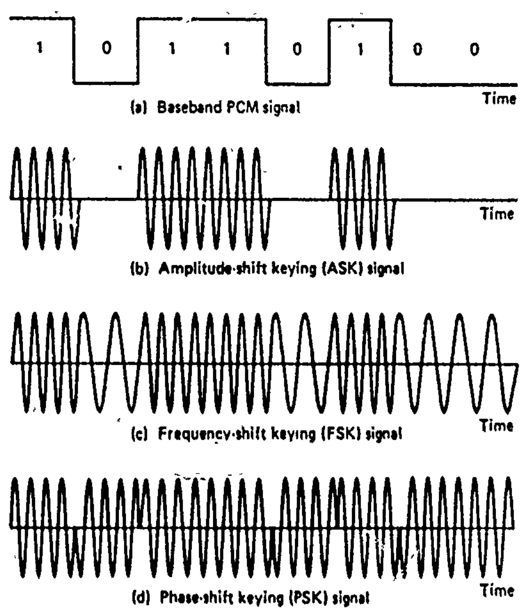
Phase Shift Keying is the last general type of digital modulation and is accomplished through the use of a fixed frequency sinusoid that changes phases abruptly. Therefore, the binary 0 would be represented by $y(t) = A\cos\omega_c t$, and the binary 1 would be represented by $y(t) = A\cos(\omega_c t + \pi)$ and can be seen in Figure 8 on page 41. A capability PSK has over ASK and FSK is the ability to make more efficient use of bandwidth by changing the phase in multiples of 90° vice 180°. Therefore, instead of just one bit, two bits are represented and can be transmitted for every phase shift. The equations are:

- $y(t) = A\cos\omega_c t + 45°$ Binary 11
- $y(t) = A\cos\omega_c t + 135°$ Binary 10
- $y(t) = A\cos\omega_c t + 225°$ Binary 00
- $y(t) = A\cos\omega_c t + 315°$ Binary 01

This process can be extended to eight different phase angles, and by using two different amplitudes, three bits can be represented. This technique is called Quadrature Amplitude Modulation (QAM). [Ref. 19: p. 75-81]

e. Bandwidth Efficiency

The bandwidth required for ASK and PSK is given by the equation, $B_t = (1 + r)R$, where r is a constant related to the filtering process and is between zero and



This figure shows the correspondence between the binary PCM signal and the resulting ASK, FSK, or PSK signal.

Figure 8. Digital Transmission Schemes [Ref. 21: p. 324]

one. The data rate is given by R. Therefore, for the worst case, B_t would be equal to twice R. For FSK, a factor of twice the frequency offset from the carrier (Δf) is added to the equation. Hence, bandwidth required for FSK transmission is given by the equation, $B_t = 2\Delta f + (1 + r)R$. However, the big advantage in digital transmission comes when multi-level signaling methods are employed, where two or more bits can be sent for every phase or amplitude shift. By comparison, ASK, PSK, and FSK bandwidth efficiencies range from .5 to 1.0, whereas QAM multi-level signaling can achieve bandwidth efficiencies of four and even five times. This means that only one fourth or one fifth of the normal bandwidth is required to transmit the signal. [Ref. 19: p. 80]

f. Detection of Digital Transmissions

ASK and FSK can be detected by processes called coherent and non-coherent detection, whereas PSK can only be detected by a coherent detection scheme. Non-coherent detection is achieved by passing the received signal through an envelop detector to extract the transmitted envelop, then filtering out undesired frequencies. After this step, the binary code is recovered from the signal and passed out as a PCM signal which is then converted to the needed form. Coherent detectors receive the input signal and use a balanced modulator to multiply the signal so that the original information signal will be one of the products. By using a low pass filter, the other products can be eliminated, and all that is left is the original information signal, but of less signal strength. Mathematically this can be seen by taking the input signal, $y_r(t) = Ax(t)\cos \omega_c t$, and multiplying by $\cos\omega_c t$. The resultant is: $y_1(t) = Ax(t)\cos^2 \omega_c t$. Using the trigonometric properties discussed earlier, this can be manipulated to look like this: $y_1(t) = A/2[x(t)\cos 2\omega_c t]$. Since x(t) is the information signal that is desired, a low pass filter can easily strip away the undesired frequencies. The resulting digital data signal can now be converted back to analog form if desired. [Ref. 21: p. 326-331]

5. Multiplexing

The telephone system uses three kinds of multiplexing: Space Division Multiplexing (SDM); Frequency Division Multiplexing (FDM); and Time Division Multiplexing (TDM). SDM is the most complex, yet easiest to describe. SDM uses individual paths for each voice circuit. Such multiplexing is expensive and is only used between the CPE and CO, or between closely located COs.

FDM is an analog signal technique of combining several channels into one frequency band that allocates separate subchannels to each subcarrier frequency. In the telephone system, the subcarrier frequencies are usually multiples of 4 KHz in a 12 channel band ranging from 60 to 108 KHz. This is the smallest FDM that occurs, and the largest FDM group can have 3600 channels with a frequency range of 564 to 17,548 KHz. As discussed in an earlier section, guardbands are needed to prevent crosstalk from adjacent channels. Figure 9 on page 43 shows how it works. [Ref. 22: p. 11.26]

TDM is the digital signaling technique used to transport many separate data signals over one communication link. The link can be a wire or a cable. The technique is accomplished by sampling the multiple inputs at regular intervals and in a precise sequence. Between each sample is placed a "dead" space that prevents crosstalk between adjacent samples. Also, recalling Shannon's sampling theorem, the sampling rate must be twice the highest frequency of the inputs to prevent signal distortion due to aliasing. Frames are then constructed using numerous input channels or sources plus one synchronization pulse. One drawback of TDM is that the overhead required from the synchronization pulse, the "dead" space, and sampling rates that are normally chosen higher than what is needed for all input channels, usually results in bandwidth requirements that are two or three times the theoretical minimum. Figure 9 on page 43 shows graphically how this is done. [Ref. 21: p. 284]

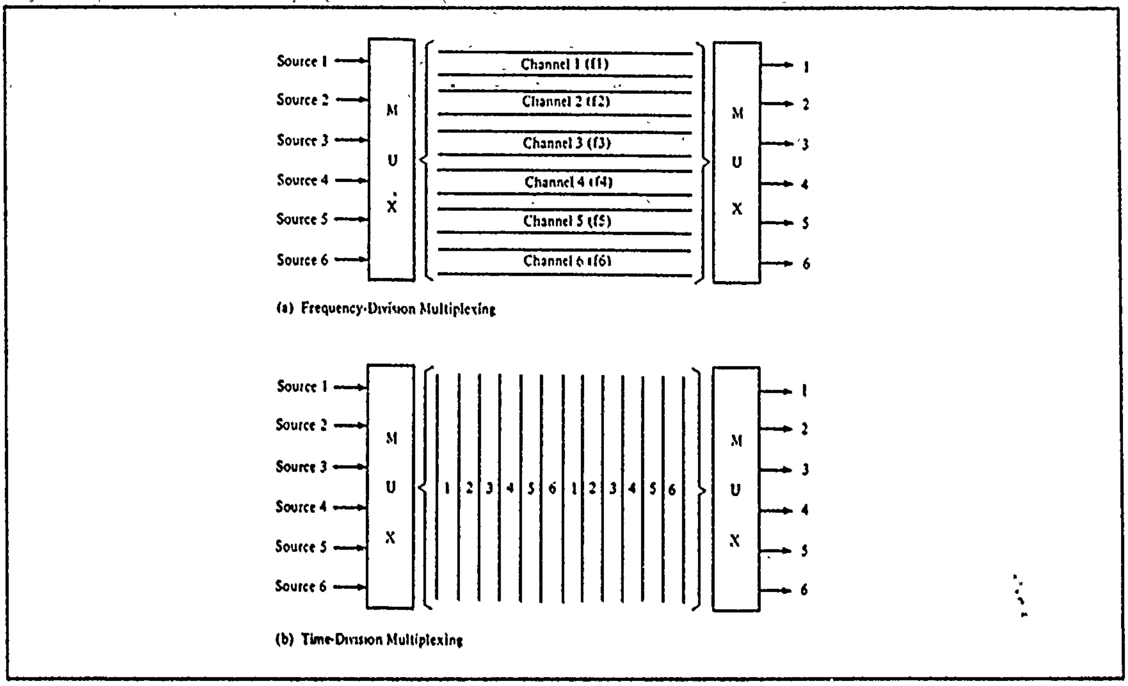


Figure 9. Frequency and Time Division Multiplexing [Ref. 19: p. 167]

Some multiplexing systems used in telephony are Digital Cross-connect Systems (DCS), channel banks, back to back, and bandwidth managers. A DCS is transmission equipment that is a multiplexer that acts like a switch and operates in conjunction with digital service levels one through four (explained later in chapter). The difference between a switch and a DCS is that switches require the incoming signal to carry destination information, whereas DCSs do not carry this information since it is carried over a separate common channel signaling system. A DCS routing matrix is established well in advance in the form of a fixed routing table. Therefore, the DCS system is considered transmission equipment since it functions more like a multiplexer than a switch. Also, most DCS systems are operated in conjunction with leased circuits. This being the case, the DCS will become the leased circuit's interface with the Public Telephone Network (PTN). The function of a DCS is to split the digital service into the appropriate number of channels and then terminate the channel at the CO, pick up a new channel, or pass the channel through to the next DCS. This function is called drop and insert and is done automatically by the DCS. Using the DCS, the various channels can be rerouted at a moment's notice by a computer controlled routing scheme. On the other hand, channel banks are used when all channels will terminate as shown in Figure 10 on page 44. A back to back multiplexer is just a manual way to drop and insert and is shown in Figure 11 on page 44. Finally, a bandwidth manager is a DCS located at a customer's premises. Another major difference in these systems is that the DCS system for the entire national network has one centralized control facility for each of the big three IXC carriers, AT&T, MCI, and Sprint. [Ref. 23: p. 38-40]

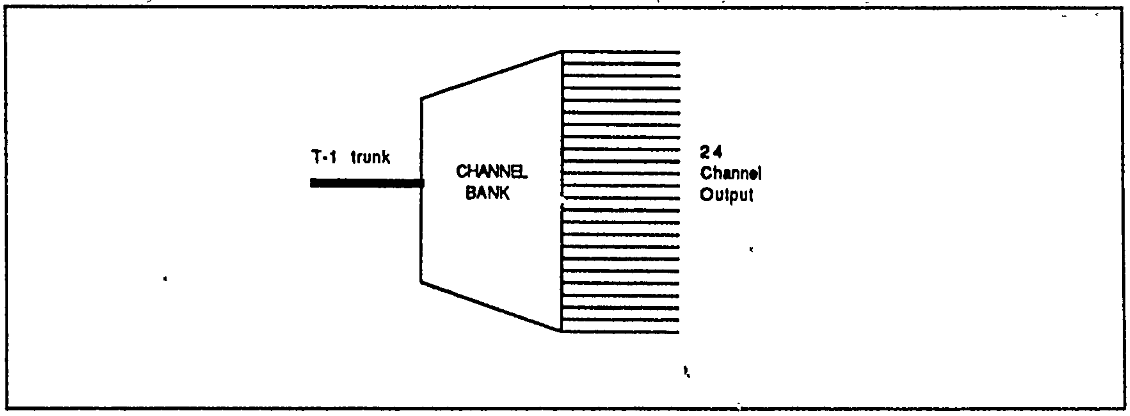


Figure 10. Channel Bank

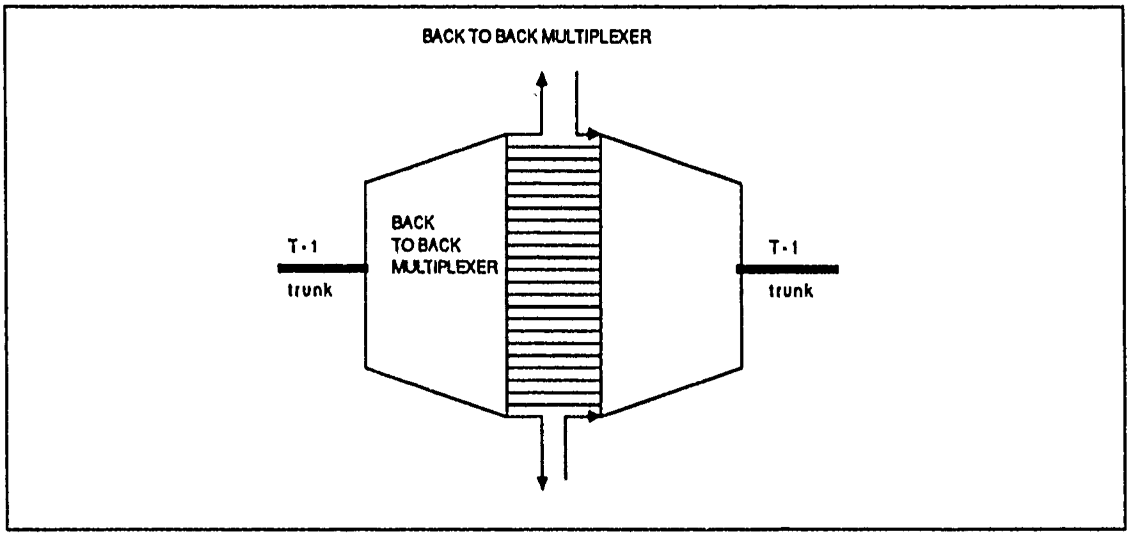


Figure 11. Back to Back Multiplexer

6. Common Transmission Problems

The two most common problems in transmission are echo and singing. Echo is when a speaker's voice is reflected back to the speaker. For this to be a problem, the reflection must endure a noticeable delay and be loud enough to be disturbing. This is usually caused by impedance mismatches in the system. Singing is the result of sustained oscillations due to amplifier feedback. Circuits that sing will quickly become unusable. [Ref. 24: p. 45]

C. SWITCHING

1. Introduction to Switching

The first question to ask is why switch? The answer is quite simple and can be seen readily in Figure 12. In the top figure, no switching is applied and results in the need for N(N-1)/2 lines to connect all the terminals. The bottom two figures show how switching can eliminate the number of lines dramatically using either a one level or two level switched network. Furthermore, as discussed earlier, multiplexing can further reduce the number of lines needed even more. However, switching is no panacea. When more switches are installed, the system becomes more complex, and with fewer lines, the problems of blocking can occur if the demand is too high and lines are overloaded. Furthermore, the network must be expandable to meet the needs of the future as well as peak traffic periods such as holidays or national emergencies.

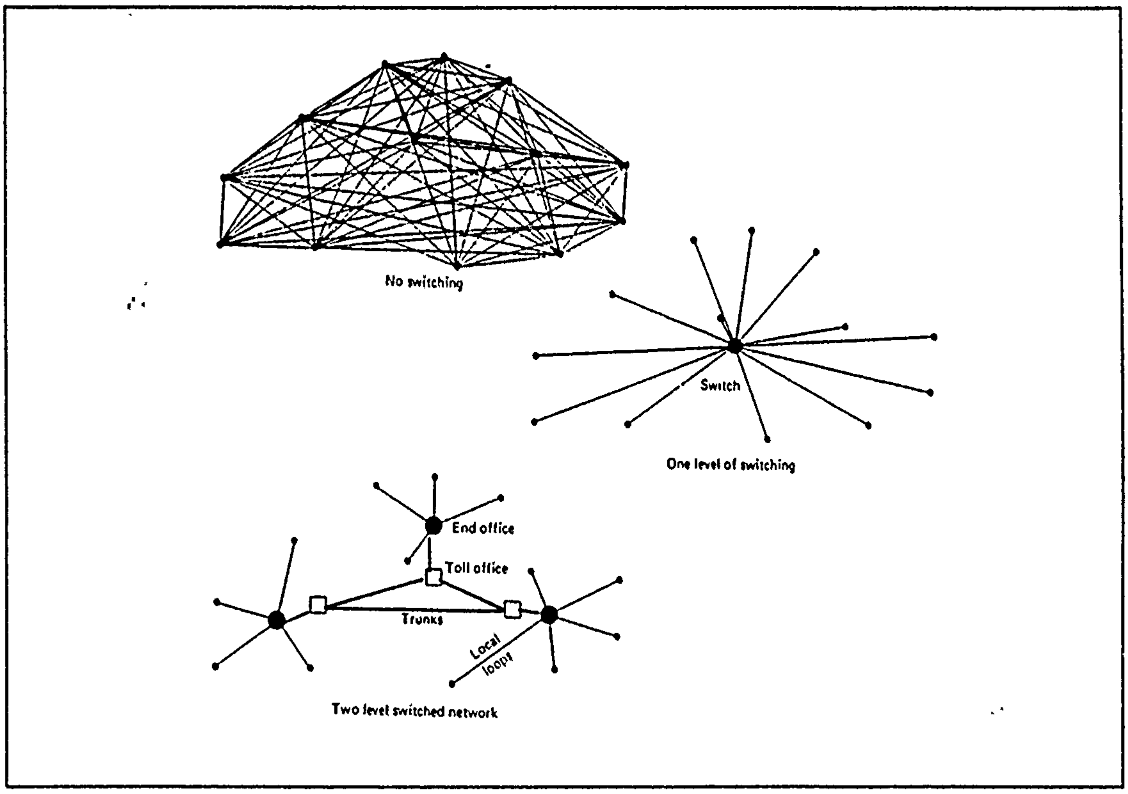


Figure 12. Effects of Switching [Ref. 25: p. 364]

In order to provide a logical and efficient means to switch, a hierarchy was established. There are two basic types of switches in the telephone system, local offices and toll offices, with the toll offices being broken down into four sub levels. The local office, end office, or CO as it is called, is the lowest switch, which is a class five switch. The end office connects the CPE to the PTN, normally via a single twisted pair. There are over 20,000 end offices in the U.S. The next class of switch is the class four and is called a toll center, followed by primary centers (class three), sectional centers (class two), and regional centers (class one). By comparison, there are only 10 regional offices in the U.S. Again, to reduce the number of lines between points, an end office cannot connect directly to another end office. The number of lines needed to connect 20,000 end offices is a staggering 199,990,000 lines! Not to mention it could get a little confusing. Therefore, if end office Y was trying to connect with end office Z, the first opportunity to switch over to end office Z's territory would come at end office Y's toll center connecting to end office Z's primary center. This is called a first choice option. If this option is not available due to loading or maintenance, the connection may be attempted primary center to primary center and is still considered a first choice option. Second choice options would be to connect end office Y's primary center to end office Z's regional center, and so on. The final choice is regional center to regional center. If this option is not available, the call is blocked. The options just described can be seen in Figure 13 on page 47.

Exceptions to this scheme are employed in large cities where it is not desirable to switch all the way up to toll or primary switching centers. To avoid this, cities employ a separate level of switching called a tandem switch or tandem office. Another method to reduce the number of telephone lines is to use devices called concentrators. A concentrator is a switching device that inputs a larger number of lines than it outputs. This is based on the assumption that all telephones from a certain geographical area will not be in use simultaneously. Based on statistical analysis, the number of telephone lines needed can be determined to meet the demand for that area. For example, in an area with a population of 80 subscribers, it is assumed that not all will be using the telephone at one time. Therefore, 80 lines are input from each CPE into the concentrator, and only 20 lines are output. This results in a savings of 60 telephone lines which is typical of the savings generated by a concentrator. [Ref. 25: p. 369]

2. Numbering Plan

In order to have effective switching, a numbering plan is needed. In the U.S., a 10 digit numbering plan is used in a 3-3-4 configuration. The first three digits are

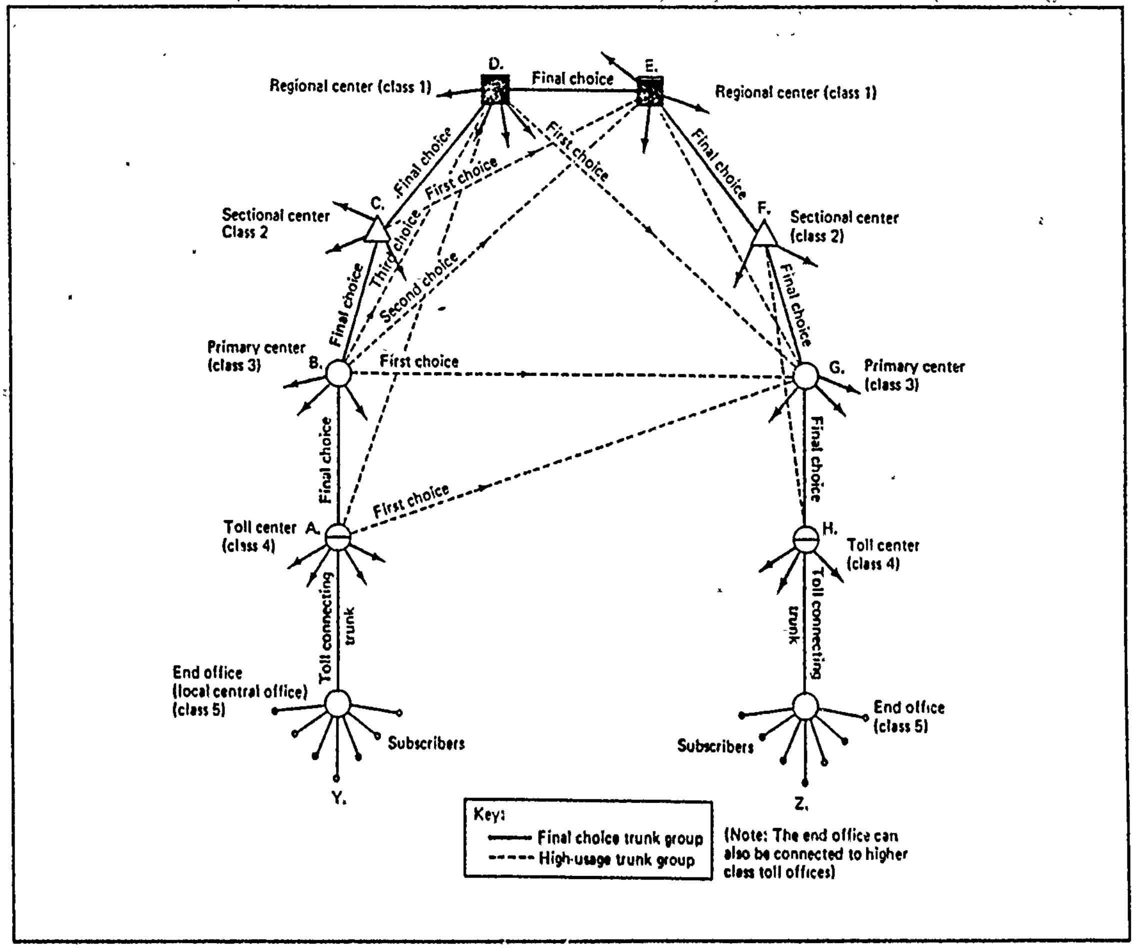


Figure 13. Switching Options [Ref. 25: p. 366].

known as the Numbering Plan Areas (NPAs) or area codes. The second three numbers are the code for the central office serving the subscriber, and the last four indicate the individual subscriber being served by that CO. Therefore, each CO is limited to 10,000 subscribers, or more if they are grouped into modules of 10,000. The area code is necessary because there are 20,000 end offices. The numbering plan routes the call to a high level toll office by use of the area code, then to the end office by use of the three digit exchange code, and finally to the subscriber by use of the last four digits. [Ref. 26: p. 109]

3. Switching Centers

Historically, switches were designed to last 40 years, which seems a bit long considering the rate at which technology is changing. With this idea in mind, the ability to expand a switching center is needed. Furthermore, reliability must be maintained near

100%. This means the use of redundant equipment and backup power supplies in the form of large storage batteries for small switches or engine driven electric generators for large switches. Also, subscriber lines or trunks enter the switch underground through a cable vault before entering the switching center at the Main Distributing Frame (MDF). The MDF is where the lines are separately connected and also protected by protection blocks, which are carbon shunt blocks designed to shunt the load in case of lightning strikes or accidental groundings at the switch. Large switching centers that have the new Electronic Switching Systems (ESS) are capable of terminating hundreds of thousands of lines, 200,000 trunks, and processing over 500,000 calls per hour. Additionally, switching centers are designed to operate without extra cooling, however, some of the more modern equipment is beginning to change this practice. All of these features cost money which is indicated by switching representing 23% of the telephone system costs in the U.S. [Ref. 26: p. 26-27]

4. The Various Switch Generations

The first type or generation of telephone switch was the Strowger switch, or step by step (SXS) switch. It was invented by a mortician in Kansas City in 1891 who was afraid the local operator was misrouting his calls. It was officially billed "the girl-less, wait-less" telephone switch. In this type of switch, a call progresses step-by-step as the telephone is dialed. The switch responds to the pulses sent to it by the rotary dial telephone or touch tone telephone. For example, using a rotary telephone, if the number four was dialed, as the dial rotates backwards, it will interrupt the DC voltage back to the CO four times, each time setting the appropriate wafer to the proper contact. If the telephone was touch tone, the multi-frequency pair would be received at the switch signaling which number was selected. Again, as each number is pushed on the telephone, the SXS moves to the appropriate contact. The switch is composed of a 10X10 matrix of three conductors. Two of these conductors provide the voice circuit and the third is for signaling. The matrix represents the 10 possible numbers of a 10 digit telephone number. As each digit is dialed, the SXS central shaft rises one step. When that digit is complete, the wipers are rotated to the proper contact. This process is repeated for all digits in the telephone number. This type of switch is used in COs and tandem offices. Disadvantages of the SXS are high maintenance costs, switching delays, noise, and no ability to obtain economies of scale in their manufacture. [Ref. 25: p. 377]

The next generation of switches is the "crossbar." This type of switch utilizes two crossbars to locate the desired connection. The two crossbars are called the horizontal select bar and the vertical hold bar. The intersection of the two bars identifies the number. Crossbar switching is faster and more reliable than SXS, but is still relatively large and not as fast or capable as electronic switching systems (ESSs). Crossbar switches are used in toll offices and COs.

The most current generation of switches are the ESSs. The electronic switch is by far the smallest, fastest and most flexible switch. The switch consists of a computer and some memory. One of the nicest features of ESS is the ability to reprogram. This is especially good when expansion is needed. Finally, the ESS has no moving parts which makes for a durable switch with very low maintenance costs. ESS switches are used in all classes of switching centers.

A system that was designed for use in the SXS and is used in all three types of switches is common control. Common control is where the entire telephone number is received by the switch before reacting. The idea is to cut down on valuable network time lost in switching. Also, switches are very expensive control devices that are only needed during call set up and take down. However, prior to common control systems, the switch was tied up maintaining the connection until the call was completed. The common control system now performs the actions necessary to maintain the call, allowing the switch to process other calls. The ESS makes the best use of this feature. [Refs. 27: p. 39-43, 25: p. 376-383]

5. Switch Characteristics

All switches have three common properties. First, they must be able to connect any input line to any output line. Second, they must have several paths available so that more than one call may be switched at a time. Third, switches are designed with the notion that not all input lines will be in use simultaneously.

In addition, switches are characterized by three different categories: space division; frequency division; and time division. Space division is just straight forward multiplexing, where any input can be connected to any output through a series of switches. In Figure 14 on page 50, it can be seen that any one of the four inputs can be connected to any one of the four outputs. Figure 14 on page 50 is an example of single stage switching. Three stage switching is seen in Figure 15 on page 51, where the first set of switches are identical to those in Figure 14 on page 50 and are then concentrated by another 4X4 switch and finally expanded to provide 16 outputs. The first two stages comprise a concentrator, as discussed earlier.

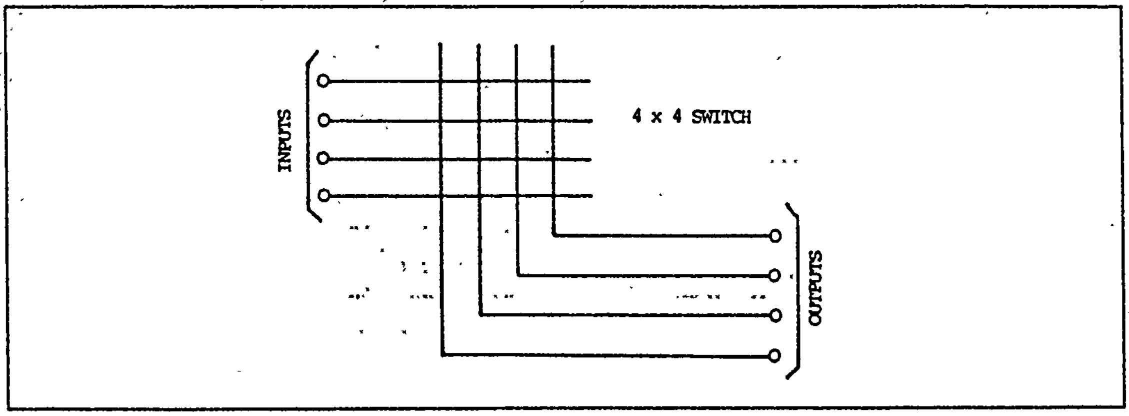


Figure 14. Space Division Switching [Ref. 24: p. 309]

Time division is rapidly becoming the most popular, since it is readily adaptable to digital systems. Time division systems are divided into Pulse Amplitude Modulation (PAM), Pulse Width Modulation (PWM), Pulse Position Modulation (PPM), and Pulse Code Modulation (PCM). The basic theory for all these types is similar. In Figure 16 on page 52, it can be seen that the common bus at the bottom carries the PAM or other type of frame from the input lines to the output lines. Low pass filters on each side remove spurious signals. The frame is put together for this example switch using one time slot per switch. If input line two (I2) is to be connected to output line five (O5), they must operate simultaneously. [Ref. 24: p. 306-328]

6. Types of Switching

There are three types of switching used today in the telephone system: circuit switching; message switching; and packet switching. Circuit switching is the most prevalent type and is accomplished by establishing a dedicated path through the network for the duration of the call. This requires the path to be set up in advance and is used for both voice and data. Circuit switching can use all three types of multiplexing, SDM, FDM and TDM. It is most efficient for calls of long duration. Message switching is used to pass data only and uses only SDM. Packet switching is the latest technology. It is readily adaptable to digital processing and uses only TDM. Packet switching is uniquely different from circuit or message switching in that the voice or data is broken down into segments called packets. These packets are then sent via the first open line toward the destination. The packets may or may not follow the same route and consequently arrive at the destination out of sequence. However, as the packets arrive, they are sequenced and processed in order. Packet switching is most efficient when the du-

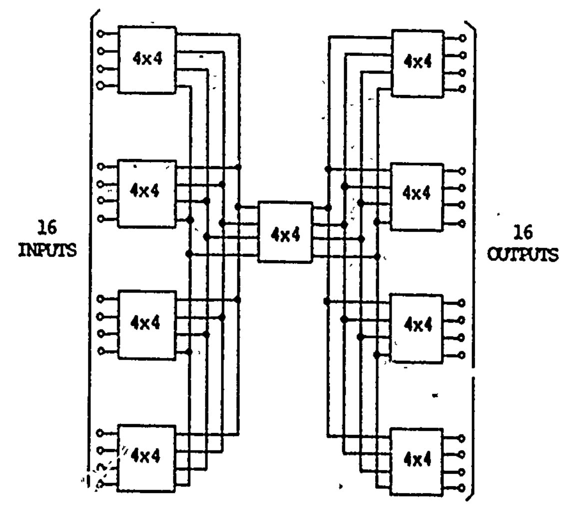


Figure 15. Three Stage Space Division Switch [Ref. 24: p. 311]

ration of the call is relatively short compared to circuit or message switching. This is because of the extra control overhead that is needed to route, packetize, and sequence each packet.

D. CONTROL SIGNALING

1. Anatomy of a Simple Call

In order to make a telephone call, there must exist extensive two way signaling between both subscribers and all switches in between. This two way flow of signals can be seen in Figure 17 on page 53, and is described below. For clarity, the diagram does not show any intermediate toll offices. When the telephone is on-hook (physically in the cradle) there exists a 2600Hz signal between toll offices that indicates there is no call. When the subscriber goes off-hook, the 2600Hz tone is broken, indicating a call is in progress. Also, the talking battery voltage drops from 48 volts DC to 5 volts DC, and a dial tone is received from the CO indicating it is ready to receive instructions. The subscriber now dials the desired telephone number that is received at the CO as a series of pulses from a rotary dial telephone or as a multi-frequency pair from a touch tone telephone. The CO interprets the number to identify the destination and locates an idle trunk by noting the presence of a 2600Hz tone. The CO records the time, date, and numbers of subscribers for billing purposes. The next toll office continues routing the call until it reaches the destination CO, which is only passed the last four digits of the

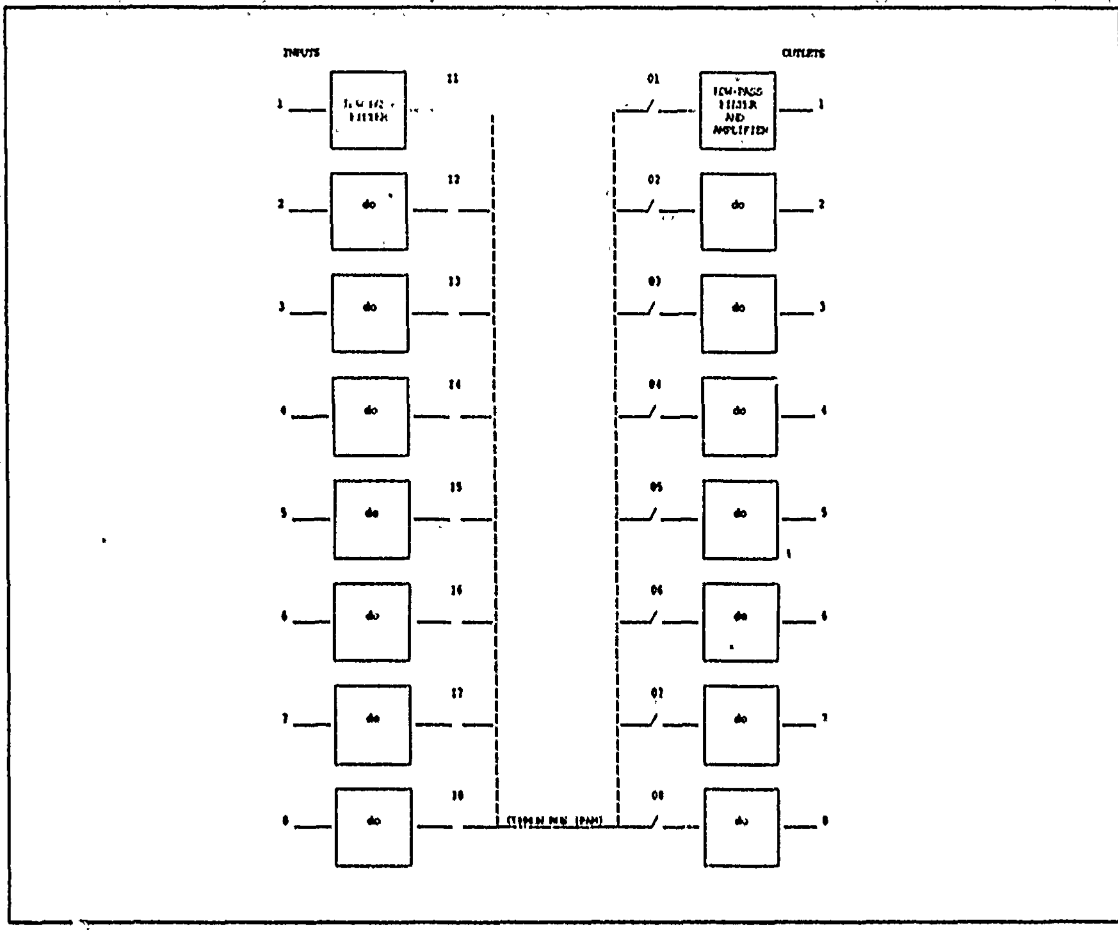


Figure 16. Time Division Switch [Ref. 24: p. 314]

telephone number. The destination CO will send a delay dial signal to the previous office until it is ready to receive, and then sends a start dial signal. Once this signal is received, the destination CO supplies 60 to 90 volts AC at a frequency of 20,000 Hz to supply power to the ringer in the destination telephone. The destination CO also sends the 20,000 Hz tone back to the caller, so the caller hears the ringing. When the called party answers, the conversation may proceed as the circuit is established. When either party hangs up, the 2600 Hz tone is resumed by the toll switches indicating they are free, and the CO disconnects the party noting the absence of the 60 to 90 VAC. [Refs. 25: p. 414-417, 18: p. 9-11]

2. Signaling Tasks

From the description of the call above, it is easy to see that a telephone switch must be capable of performing the following tasks: signal reception; signal interpreta-

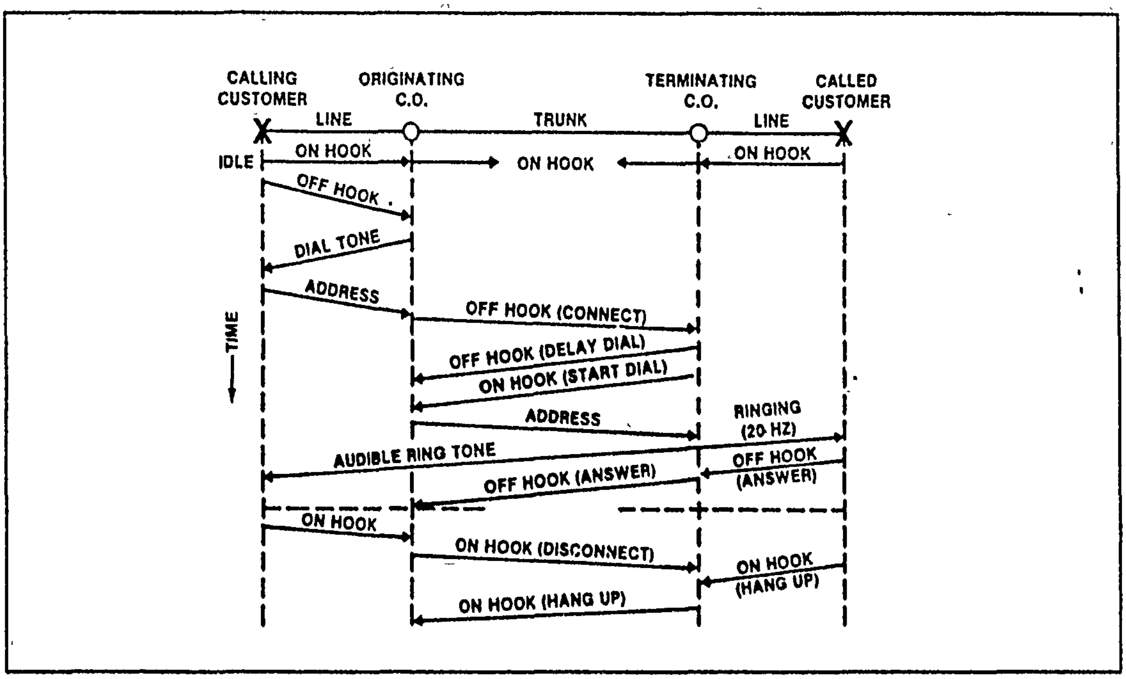


Figure 17. Signaling Required for a Normal Telephone Call [Ref. 26: p. 101]

tion; storage; path selection; and signal transmission. Signal reception is the means by which the signal is received from the subscriber or previous switch and includes touch tone, multi-frequency, DC loop closure, etc. Signal interpretation is where the signal from the subscriber is interpreted into a switching plan using the programmed matrix of the switching device. Storage is necessary primarily as a buffer for signaling information, retention of call status, and translation information needed to convert directory numbers to equipment numbers such as trunk numbers. Path selection is determined by trial and error until a free path can be identified. Finally, signal transmission is where the signal is transmitted between switches or subscriber and end office. [Refs. 28: p. 13, 17, 24]

The circuit that handles all of the requirements discussed above is called th BORSCHT circuit which is short for Battery, Overvoltage, Ringing, Supervision, Coding, Hybrid, and Testing. The battery is the talk battery located at the CO that supplies the 48 volts DC necessary for the telephone instrument to signal by dialing and also the five volts DC needed to power the carbon microphone. Overvoltage is to protect against lightning strikes or accidental groundings to power lines. Ringing is the 20,000 Hz signal that alerts the called party and the caller that the telephone is ringing at the destination end. Supervision is the monitoring of on or off-hook through sensing the presence or absence of AC current. Coding is used when a CODEC is used to convert back and forth between analog and digital signaling. Hybrid is the circuit that separates incoming and outgoing signals. Finally, testing is the metallic access provided to a line for periodic testing. [Ref. 28 : p. 237-241]

3. Supervisory and Address Signals

Signaling can be thought of as a subset of switching where the signaling is controlling the switching. The process of signaling may require decoding, encoding, signal generation, and transmission depending on the signaling system's purpose. In a sense, signaling systems are a communication system within a communication system that transports the information necessary to route calls and control the network. Signaling is accomplished by passing signals between each successive switch rather than end to end signaling. This requires the signal to be received and interpreted at each switch. Finally, signaling is accomplished by AC and DC voltages sent in both directions. [Refs. 24, 25]

Signaling is divided into two broad categories, supervisory and address. Supervisory signals are comprised of control and status information. Most control states are usually binary in nature. Examples of control signals are on-hook and off-hook signals. Status signals have more than two states and include busy and ringing signals. Address signals are usually either DC pulses from a rotary dial or multi-frequency tones from a touch tone telephone. Supervisory and address signaling can be further broken down into audible signals and network management. Audible signals are exactly like the name implies and include ringing and busy signals. Network management signals take care of routing calls to optimize network utilization and avoid congestion. [Ref. 24: p. 328-340]

4. Signaling Functions

The list below is a representation of what signaling provides. [Ref. 25]

- Audible communications with the user: dial tone, busy signal, ringing, or other to indicate something is malfunctioning
- Transmission of telephone number to switches
- Communication between switching centers if call cannot be completed
- Disconnect signals between switching centers
- Signal to make telephone ring
- Billing signals
- Network status signals used to alert maintenance crews and for the network to avoid congestion

- System diagnoses
- Control of special equipment such as satellite channels and Time Assignment Speech Interpolation (TASI)

TASI is used primarily on expensive overseas cables to increase the efficiency of the system. During the course of a normal conversation, audible speech is present only 45% of the time. By using TASI switching equipment, the remaining 55% can be utilized by disconnecting the call from that circuit when no speech is present and allowing another call to use the circuit. When speech is detected on the original circuit, another circuit that is vacant provides the connection. TASI improves line efficiency from 45 to 80%. [Ref. 25: p. 406 and 653]

5. Functional Areas of Signaling

The two functional areas of signaling are categorized as signaling in the subscriber loop and interexchange signaling. Subscriber loop signaling consists of off-hook (DC), dial tone (AC), dialing signals (DC pulse or Multi-Frequency (MF)), busy tone (480 or 620 Hz AC), and ringing tone (20,000 Hz AC). All of these have been explained in previous sections, except MF, which is quite simple. MF signaling is where two separate tones are produced by depressing one single button on a touch tone telephone. As seen in Figure 18 on page 56, each button has two frequencies assigned to it. Every time that button is depressed, those two frequencies are transmitted to the switch and interpreted. Interexchange trunk signaling is the signaling between offices or switching centers that can be categorized as out-band, in-band, or common channel signaling.

a. Out-Band Signaling

Out-band signaling is the least common and used only in signaling system number R2 (SSR2). The concept of out-band signaling requires that all signaling be transmitted over a separate and dedicated line devoted to signaling. The primary advantage is that out-band eliminates interference with the voice or data information. The primary disadvantage is that it requires a complete, separate network, which usually costs more and provides limited bandwidth. In the case of SSR2, the signaling that occurs is essentially the same as the 2600 Hz in-band signal but is sent over a separate network at a frequency of 3700 Hz. [Refs. 25: p. 412, 26: p. 106]

b. In-Band Signaling

In-band signaling is very common in today's telephone network. In-band signaling is exactly opposite of out-band signaling and carries the voice or data information in the same channel as the signaling information. A good example of in-band

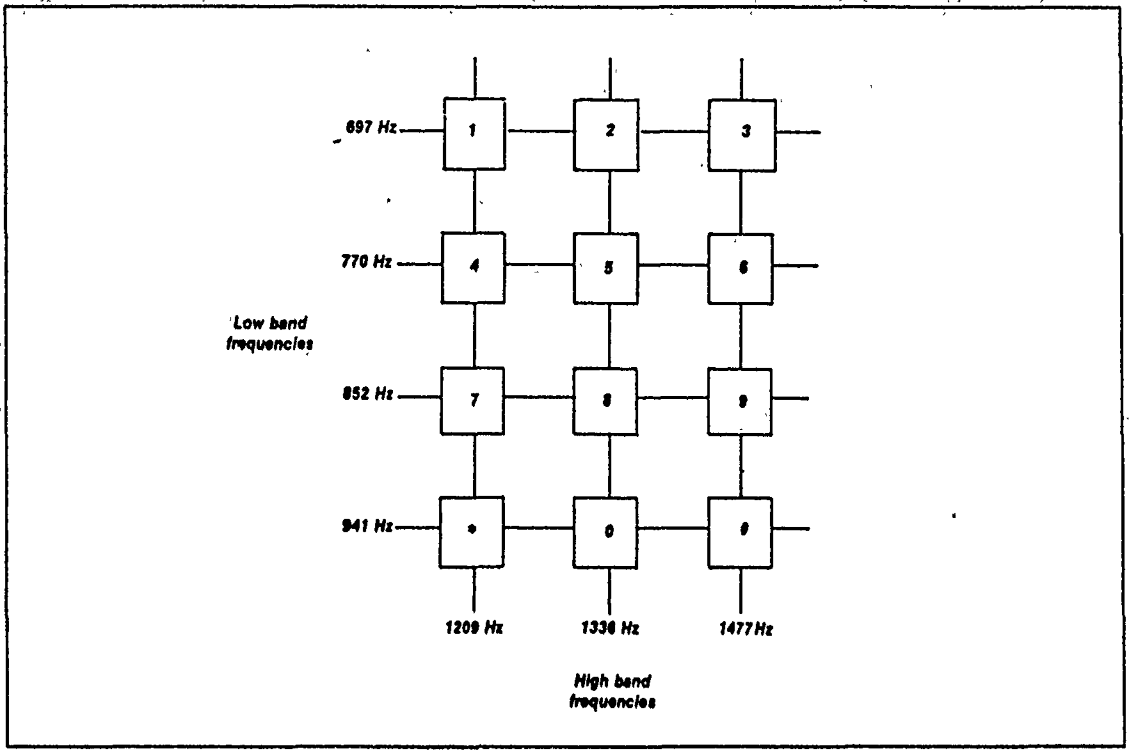


Figure 18. Multi-frequency Signaling from a Touch Tone Telephone [Ref. 18: p. 15]

signaling is robbed bit PCM signaling. In digital telephony, there is a standard called T-1, which equates to a specified capacity of 1.544Mbps. In order to transmit the digital information, a frame called a D4 frame is made. The D4 frame consists of 24 time slots or channels with eight bits per slot, and one framing bit (F-bit), for a total of 193 bits per D4 frame. The F-bit is used for frame synchronization. The D4 frame is repeated 8,000 times per second giving a data rate of 1.544 Mbps. Twelve D4 frames are grouped into a D4 superframe where the least significant bit in every sixth frame is "robbed" and overwritten to encode the control signals. Thus each channel will have 2 signaling bits allowing four signaling conditions. These signaling conditions are the signals passed between offices, such as a busy signal. Because the least significant bit is "robbed," this means that one out of every eight bits cannot be used for data, since the change in one bit will give the binary word a whole new meaning. Therefore, the effect of robbed bit signaling with respect to data transmission is to reduce the available capacity per channel to 56 Kbps vice 64 Kbps available for voice communications. This is because voice communications are more tolerant of noise and will still be intelligible if only the two bits are overwritten. Also, an Extended Superframe (ESF) version is being used, which uses 24 D4 frames. ESF achieves this by robbing one bit every six frames for a total of four bits, which allows 16 signaling conditions. The advantage of in-band signaling is that the signal and information are transmitted over the same network. The disadvantage is caused by signal and information interference that can lead to erroneous and undesirable signals. [Ref. 20]

c. *Common Channel Signaling*

Common Channel Signaling (CCS) is similar to out-band signaling, where a separate network or channel is used to transmit the signal. However, unlike out-band signaling, CCS allows multiple signals for many channels to be multiplexed onto one channel and that channel may take a completely independent route that does not parallel the information as in in-band signaling. In addition, CCS can transmit network management and maintenance information which allows the switching system to update its routing tables to avoid congestion and system faults or accommodate system expansion. CCS can also look ahead in the network to see if the termination point is busy before it seizes a trunk. This way it optimizes the utilization of the trunk. Another problem CCS systems can solve is "glare." Glare is when a line is seized by both ends simultaneously. The only disadvantage of CCS systems comes from the fact that the signal travels a separate route and therefore never checks the route the information must pass. To eliminate this problem, a tone generator and detector is connected momentarily to the trunk as a test. The advantages of CCS are numerous and are listed below. [Ref. 29: p. 31]

- Increased signaling capacity and capability
- Faster signaling with fewer delays
- Eliminates talk-off (phenomenon where speaker or spurious noise is produced at a frequency of 2600 Hz that causes the toll offices to sense an available line and the call is terminated prematurely)
- Reduces equipment needed

The two most popular CCS systems are CCS6 and CCS7. CCS6 is a circuit switched analog system that was designed to provide signaling between electronic switches. The signaling data are built from 28 bits which include eight error detection bits. These bits are passed using a modem operating at 2400 Kbps. The CCS7 system is similar to the CCS6 system, except that it is an all digital system designed to work with the newer all digital communication equipment. Figure 19 on page 58 depicts the CCS7 system which is built around a 56 Kbps packet switched network. The CCS7 network components consist of Service Switching Points (SSP), Signal Transfer Points (STP), and

Signal Control Points (SCP). SSPs are switches located at the CO or tandem office that interface the CCS7 system to the telephone network. STPs are packet switches for the CCS7 system, and SCPs are the processors that control the CCS7 system. The CCS7 has resulted in a six percent increase in throughput over the CCS6 system and offers additional services that include improved 800 service, credit card calling, and Custom Local Area Signaling Service (CLASS). CLASS enables features like call forwarding. However, the major improvement is increased speed and capacity over previous systems.

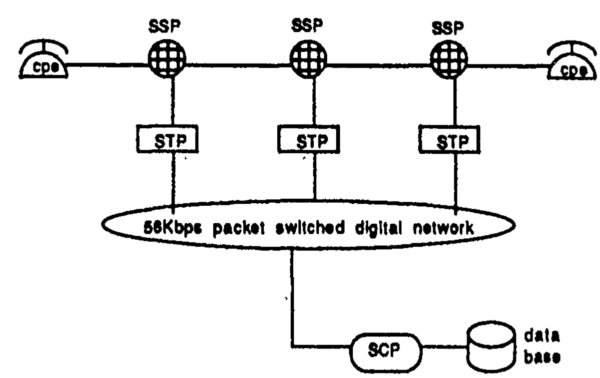


Figure 19. Common Channel Signaling System Number 7 (CCS7)

A point that needs to be emphasized is the speed of the CCS systems, particularly the CCS7 system. Recall that telephone design relies on analysis of telephone traffic that is measured in erlangs. This analysis is used to determine the number of trunks, lines, and equipment needed for that portion of the telephone network. In-band signaling typically takes 15-20 seconds to establish a call compared to 1-2 seconds for the CCS7 system. Since call set up adds to the length of the call, the number of erlangs is increased over the network. Therefore, by reducing the call set up time, the number of erlangs can be reduced, which will require fewer trunks and less equipment. This results in a significant cost savings, not to mention a more efficient network. [Ref. 20]

6. Private Branch Exchanges, Centrex, and Key Telephone Sets

Early telephone switching centers were manual plug in type where the operator physically connected the originator to the destination. As companies grew and their telephony needs increased, it became more economical to install switches on the company's premises which became known as Private Branch Exchanges (PBXs). The economy was achieved by reducing the amount of capacity and switching that was required by the CO. This was achieved by the PBX being able to switch all intra-company calls on the premises and not having to route calls to the CO for switching. As switching technology advanced, so did PBXs. By automating their switching functions, PBXs became known by some as Private Automatic Branch Exchanges (PABX). The first digital PBX was manufactured by Rolm in 1981. [Ref. 30: p. 1.5-1.6]

Figure 20 shows the four major components of a PBX. The line interface is the device that connects all the telephone terminals or computers that can either send voice or data to the switching matrix. The line interface also connects to the control unit that analyzes the control signals (MF, dial pulses, etc.) to instruct the switching matrix what route to establish. Once a path is established, the voice or data flows through the switch and through the trunk interface device to the appropriate trunk. The trunk interface functions like the line interface for calls coming into the PBX from outside the premises. These calls can originate from the local TELCO CO, a leased T-1 line, or private company network. The four components are the same whether the PBX is analog or digital. The difference is that the digital PBXs incorporate digital switching methods. Nearly all the new PBXs being manufactured today are digital and capable of switching data and voice in a mixed environment. The reason the trend is toward digital PBXs is that they can switch voice more economically than analog switches. [Ref. 30: p. 2.1-2.10]

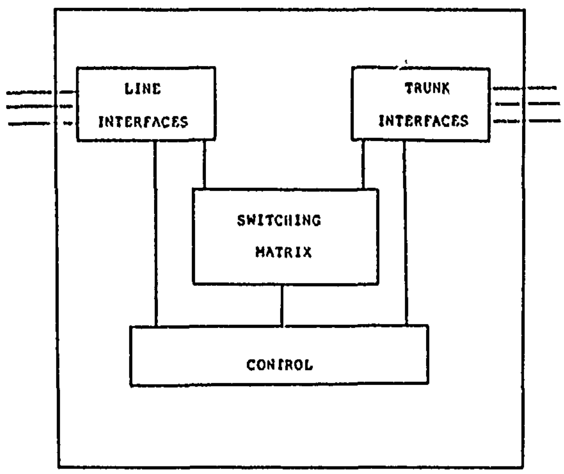


Figure 20. Major Components of a Private Branch Exchange [Ref. 30: p. 1-9]

Centrex systems are essentially the reverse of PBXs. Instead of the PBX on the company's premises, the functions of the PBX reside in the local TELCO's CO. At first glance, this does not seem economical because it requires each telephone or communication device to be wired individually to the CO. For the customer, economy is a tradeoff between the high initial capital investment in a PBX versus only the monthly service charge for Centrex. Also, the TELCO must provide equipment that is compatible with state-of-the-art communications and precludes a company from purchasing a PBX that may be obsolete within several years. Centrex allows the company to expand capabilities more freely and does not "lock" it into a long term investment. Other advantages of Centrex include TELCO-provided backup equipment, lines, and power supply in the event of failure. Also, maintenance is the responsibility of the TELCO with a Centrex system, whereas PBX maintenance is the responsibility of the company.

Key Telephone Sets (KTS) are essentially very small telephone switching centers located within the telephone terminal. They may either have several buttons or "keys" so that a different telephone line can be selected, or they may have a two position rotary switch. The most common type is the six button, but large offices may have up to 30 buttons. [Ref. 27 : p. 56-57]

E. MEDIA

1. Selection of a Medium

Before a medium can be selected for the particular application, the requirements of the system must be known. The basic media available for use in the telephone system are twisted pair, coaxial cable, fiber optic cable, microwave and satellite. Each of the media has strengths and weaknesses and should be selected only after analyzing bandwidth (capacity) requirements, modulation schemes, length of line, allowable error rates, terrain, weather, and all life cycle costs.

For short haul applications, twisted pair has been the predominant medium and is installed between the CO and CPE in most applications. For long haul applications, microwave and coaxial cable have been the most extensively used. However, fiber optics costs have been dramatically lowered in the past few years and has made fiber optics the new medium of choice in practically all applications where a cable medium was used, including long haul and short haul. The only exceptions to this are certain microwave applications and long haul satellite applications where these systems retain an advantage. [Ref. 22: p. 9.1-9.6]

2. Determining Bandwidth for a Single Channel

After careful analysis conducted by Bell Labs, the typical telephone circuit was designed to carry voice between the range of 300-3400 Hz. By analyzing Figure 21 on page 62, it can be seen that the normal hearing range for a person is approximately

20-20,000 Hz with faithful reproduction of music needing only 45-12,000 Hz. To reproduce speech without distortion and to clearly understand the speaker requires even less. The reason behind this large reduction in bandwidth can be better described by looking at Figure 22 on page 63. In this figure, the intensity level of the speaker's voice is represented by the ordinate and the frequency is a logarithmic scale along the abscissa. Notice that the minimum intensity level occurs at approximately 3400 Hz. This is the reason the upper frequency limit was chosen to be 3400 Hz. Furthermore, the bandwidth need not extend all the way to zero since attenuation distortion increases rapidly at frequencies below 300 Hz. Therefore, frequencies outside of the 300-3400 Hz range do not significantly add to the percent articulation, which is the percentage of sounds correctly interpreted by the listener. The information in this range is often referred to as the emotional content. [Refs. 22: p. 10.10-10.14, 24: p. 276]

3. Twisted Pair

When the telephone network started to grow, the medium in use was open pair wires. The name reflects the fact that no insulation jacket or sheathing was used. The open pair wires had many drawbacks due to their simple design. The most common problems were crosstalk caused by wires strung too closely together and attenuation caused by environmental factors. Rain caused leakage at the insulators, and high temperatures caused the line's resistance to increase, which increased attenuation even more.

Due to these problems, a better medium was sought, and twisted pair was the result. Because the twisted pair wires are insulated by a non-conducting jacket and the wires are twisted together, the problem of crosstalk was minimized. The frequencies on adjacent wires are chosen to specifically minimize crosstalk, and the twist rate will vary for different frequencies. One disadvantage was that twisted pair wires used smaller diameter wire, which created higher resistances that required amplifiers to be installed every three to four miles vice every 10 miles for open pair wires. As is the case with all twisted pair transmissions, the information frequency is raised by the carrier to a higher transmitted frequency which increases the effects of attenuation, distortion, and capacitance. However, raising the frequency allows the use of FDM. [Ref. 25: p. 156-164]

The phenomenon associated with higher frequencies attenuating faster is called amplitude distortion. The distortion is caused by lower frequencies being received at stronger levels than the higher frequencies, and the effect is to distort the speaker's voice. The effect over a long route at high frequencies can be devastating to the signal. The method used to defeat this problem is to try and equalize the loss for the frequency range

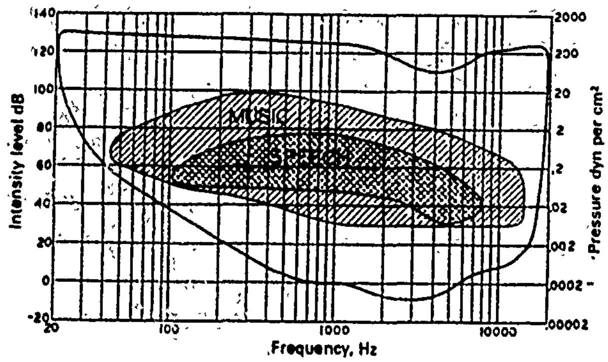


Figure 21. Audible Frequency Range [Ref. 22: p. 10.10]

by inserting tuned loading coils into the lines at periodic intervals approximately one mile apart. Figure 23 on page 64 shows what the difference is between unloaded and loaded lines. Notice the sharp rise in attenuation at the higher frequencies. [Ref. 31: p. 20]

Another problem is delay distortion caused by the differences in propagation velocities at different frequencies. Since higher frequencies have more resistance, they will have higher delay distortion. The effects of delay distortion are considerably worse when the information is data rather than voice. Once again, loading coils are used to reduce this problem. Figure 24 on page 64 shows how the velocity curve for an unloaded pair changes rapidly , whereas a loaded wire pair has only a modest increase over the selected bandwidth.

Today, twisted pair is still used in sizes ranging from 19 to 26 guage. The heavier 19 guage wires are used in long trunks to keep resistance low and need for amplification down. The 24 and 26 guage is the size chosen to connect the CO to CPE. Depending on the size, twisted pair cables are available in bundles of six to 2,700 pairs. As a rule, the resistance is limited to 1300 ohms which limits the CO range. For a comparison, 1300 ohms equates to 15 miles using 19 guage whereas 26 guage equates to only three miles. This is why longer routes use heavier wires. [Ref. 24: p. 285-286]

4. Coaxial Cable

As discussed earlier, twisted pair has some limitations caused by limited bandwidth and distortion. These limitations are functions of the wire diameter. In order to get a larger bandwidth, higher frequencies must be used which increases delay and amplitude distortion. The longer the route, the worse the problem gets. In order to solve

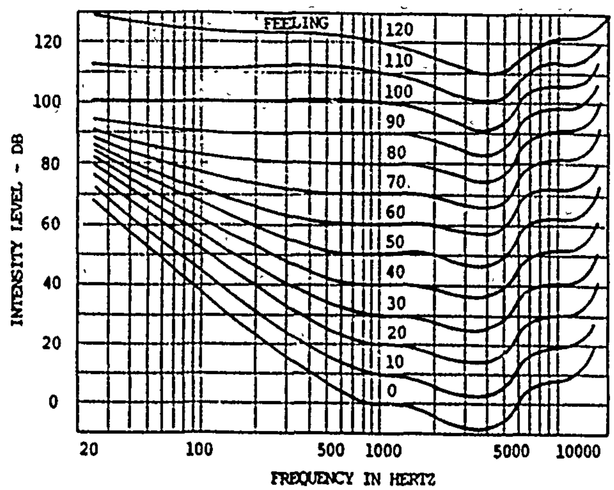


Figure 22. Sensitivity of the Ear vs. Frequency [Ref. 24: p. 276]

these problems, the wire must be larger in diameter. The problem is that a solid copper wire of sufficient diameter is too costly. However, engineers were able to take advantage of a phenomenon called skin effect to solve these problems. Skin effect is the tendency for electrical current to flow on the outer wall of a conductor and the tendency increases with frequency. Therefore, the logical way to reduce cost while increasing capacity was to create coaxial cable. The coaxial cable is made from two conductors. The outer shell is where the information flows. It is a hollow cylinder, usually made of copper, that surrounds a single wire that supplies the source of power. The voltage of this power source may be as high as 4000 ±2000 volts for the largest cables. The material between the two conductors is usually plastic or just an open air space. By taking advantage of the skin effect at higher frequencies, a trade off in cable diameter was possible that allowed the coaxial cable diameter to decrease. Typical coaxial cables carry 3600 voice channels all the way up to 13,200 channels. Furthermore, cables are bundled together so that as many as 132,000 two way voice channels are contained in one cable. Advantages of coaxial cable over twisted pair are the increased capacity, decreased crosstalk potential, lower delay distortion, lower amplitude distortion, and higher propagation speeds. Higher propagation speeds are possible mainly because loading coils are not necessary with coaxial cables. Typical propagation speeds of twisted pair wire are 10,000 to 20,000 miles per second, whereas coaxial cables propagate the signal at speeds up to

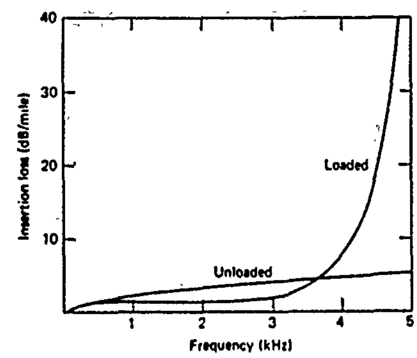


Figure 23. The Effect of Loading Coils on Amplitude Distortion [Ref. 31: p. 20]

10 times that speed depending on the insulating material used between the conductors. [Refs. 25: p. 162-166, 24: p. 299]

5. Fiber Optics

The idea of light waves carrying communications is not new. As a matter of fact, Alexander G. Bell pioneered light wave communications in 1880 when he invented the photophone. The photophone was a device that used reflected sunlight to transmit the sound to a receiver. Bell considered this his greatest achievement. As time passed,

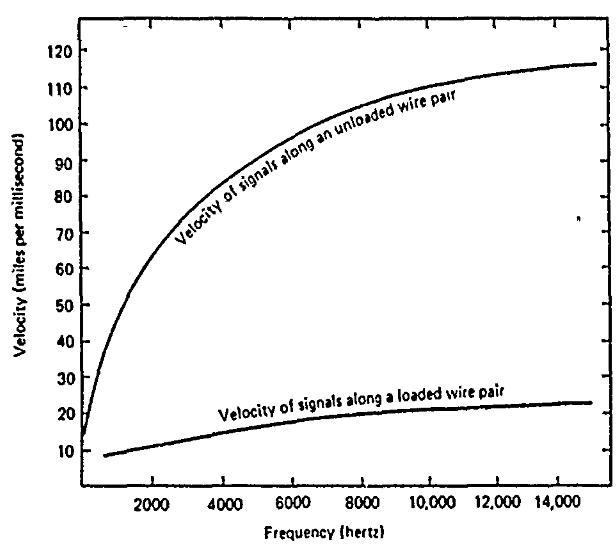


Figure 24. Effect of Loading Coils on Delay Distortion [Ref. 25: p. 165]

several inventors continued this basic idea attempting to use glass fibers as the medium to direct the light source. The main problems were the lack of a practical, high intensity light source and the ability to manufacture glass at low impurity levels. The invention of the laser (Light Amplification by Stimulated Emission of Radiation) in 1958 by Bell Labs provided the prospect of a practical light source. However, the best quality glass at the time would attenuate light signals to undetectable levels in a matter of only a few meters. In 1970, spurred by the invention of the laser, Corning Glass succeeded in making a silica glass of sufficient purity to be a practical medium. Simultaneously, Bell Labs invented the heterostructure optical device which was a semiconductor laser using Gallium-Arsenide. This device provided an improved light source that could operate continuously at room temperature. [Refs. 32: p. 1-20, 33: p. 26-40]

Today, optical fibers are composed of three sections called the core, cladding, and jacket. The central core and cladding are made of very pure plastic or glass with the central core having a slightly higher index of refraction then the cladding. This is because fiber cables use the principle of Snell's law to bend the light. By adjusting the indexes of refraction for the core and cladding, total internal reflection can be achieved at the boundary of the two materials. The outer jacket provides protection from nicks, cuts, and external light interference. [Ref. 34: p. 1-12]

There are three types of fiber optic cable, each having unique characteristics, as shown in Figure 25 on page 67. The first type is step-grade multimode which is formed by two distinct indexes of refraction (n). As seen in Figure 25, several paths are taken by the light which results in a time of arrival difference due to the longer path the steeper rays must take. The net result of this causes the received pulse to be spread or dispersed and is called modal dispersion. In order to reduce modal dispersion, another type of fiber optic cable called Graded Index multimode (GRIN) is used. GRIN fiber uses a parabolic shaped index of refraction (n) curve for the inner core. This slow change in n allows the different paths to arrive within a very close time interval. This is true because the outer light ray paths are longer, but travel through regions of lower indexes of refraction and propagate faster. The overall effect is that the propagation delays of all paths are nearly equal. The third type of fiber optic cable is single mode where there is only one propagation path, and modal dispersion is not a factor. However, chromatic dispersion is a factor and is the result of different wavelengths of light propagating at different speeds. Chromatic dispersion can be subdivided into two components: material dispersion and waveguide dispersion. Material dispersion is caused by the variance in n with respect to wavelength. Waveguide dispersion is a function of core dimensions and wavelength. Therefore, chromatic dispersion depends on the properties of the fiber as well as the light source frequency spectrum. On the left side of Figure 26 on page 68, it can be seen that minimum signal loss per kilometer is achieved at source wavelengths of 1.3μm and 1.55μm depending on the type of fiber cable in use. Now turning to the right hand side of the figure, the case of a single mode fiber is represented by line one. It can be seen that chromatic dispersion goes through zero at 1.3μm and increases rapidly for wavelengths on either side. The significance is that for wavelengths of 1.3μm, the signal loss and chromatic dispersion are both minimized. If desired, the fiber material can be doped to shift the dispersion curve to zero at 1.55μm to achieve the same effect. This is seen in line two. New materials are being tested that have flat dispersion curves over a wide range of wavelengths with much less loss per kilometer as demonstrated by line three. Chromatic dispersion affects all three types of fiber but is not the primary concern for multimode fibers where modal dispersion is the dominant dispersion problem. [Ref. 34: p. 1-12]

Although the glass fibers are stronger than steel with respect to tensile strength, their small size requires careful handling. Typically, fiber optics cable structures range from loose tube designs, with individually buffered fibers, to ribbon cables in which the fibers are sandwiched between two strips of plastic. The last type of structure allows a higher packing density. Today's fiber optic cabling systems have channel capacities of 23,040 channels per cable using repeater spacing of up to 10 miles. The highest capacity coaxial cable is L5E. It only carries 13,200 voice channels, and repeaters must be spaced every mile. [Refs. 22, 25, 34]

Included in the fiber bundle will be some form of strength member to alleviate stress and strain on the fibers. The material is usually braided from steel, copper or kevlar and may form the core of the bundle or may lie just under the protective outer sheathing made out of plastic. Also, fiber cable bundles contain copper or other metal wires normally incorporated into the outer sheathing that are used to supply power to the repeaters. In some cases, the strength members and repeater power supply wires will be the same. If the fiber cable is part of a main trunk, the repeaters may have backup power supplies using batteries that are trickle charged through normal system use. These batteries can last for several days. [Ref. 35: p. 51-53]

Signal degradation can occur through several other means. The easiest to prevent is to limit the bending of the cable to reduce stress fractures in the glass. The most serious problem is caused by poor splicing. Splicing is accomplished by joining the two cable ends by fusion or bonding. The alignment of the cable is critical and is accom-

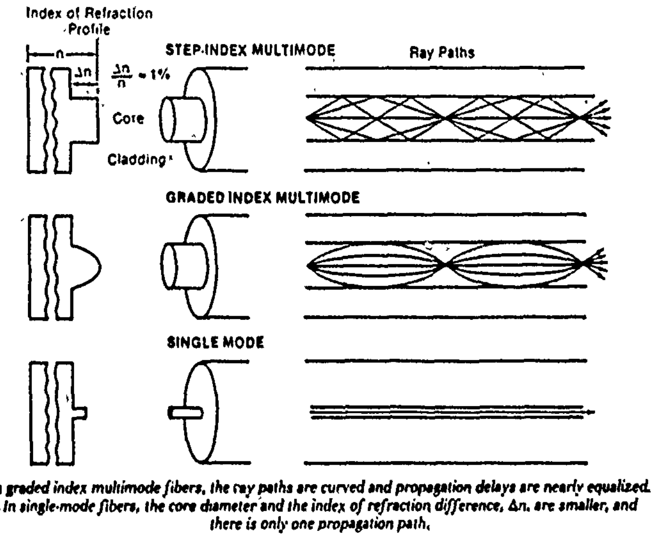
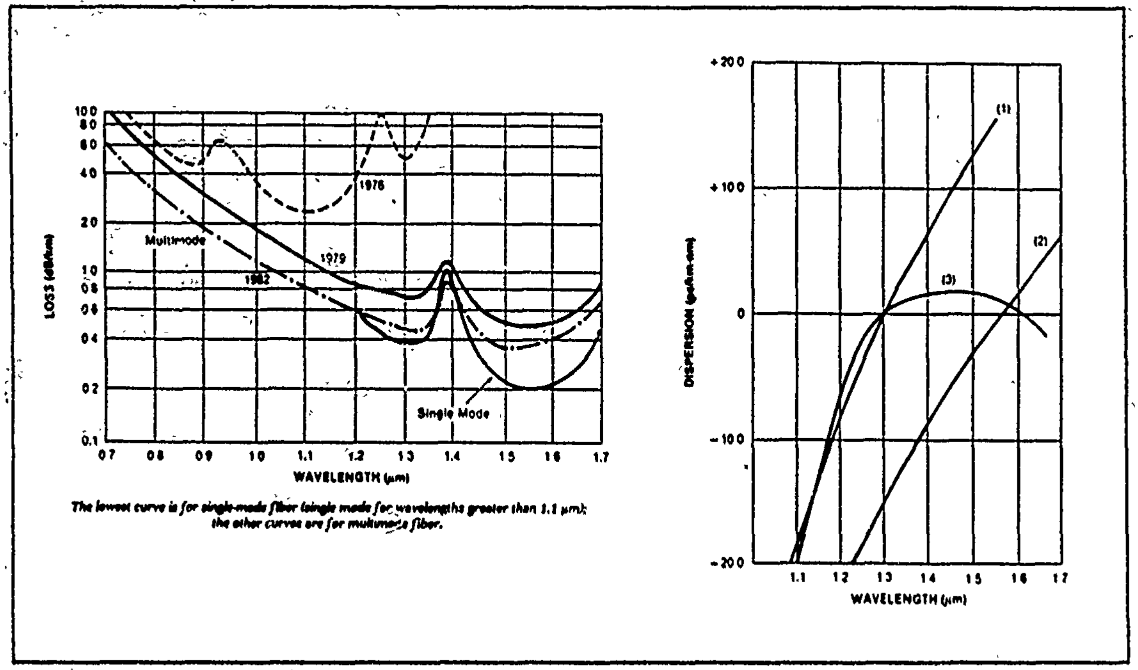


Figure 25. Types of Optical Fiber [Ref. 34: p. 9]

plished by an optical measuring device that is sensitive to within a micrometer. If the ends to be joined are rough, misaligned angularly, or offset, a loss will occur proportional to the deviation. As seen in Figure 27 on page 69, the effect of offset misalignment is greatest in single mode fibers. Notice that shortly after three μm offset, the single mode fiber insertion loss rapidly rises to infinity. [Ref. 34: p. 13-14]

The source of light for the fiber cable comes from either an Injector Laser Diode (ILD) or a Light Emitting Diode (LED) with each type having distinct advantages. The LED is by comparison inexpensive and very reliable. Its disadvantages are that the light emitted by an LED is spread over a wider spectrum range resulting in increased chromatic dispersion, and the light is not as focused which results in much weaker intensity levels. By comparison, ILD light sources provide a focused beam of light over a narrow spectrum and can be modulated at a higher rate. However, the disadvantages are high cost and heat that is generated by the laser that must be dissipated. Regardless of light source type, the source will be modulated by externally changing the light after it leaves the ILD or LED, or the light source will be directly modulated. Normally the input current is modulated in one of three ways: directly modulating the input current that creates an intrinsic time delay needed to bring the energy level up to the threshold to emit light; biasing the light source to just below the threshold to minimize delay (although this causes more noise); and biasing the light source to just above the threshold



The lowest curve is for single-mode fiber (single mode for wavelengths greater than 1.1 μm); the other curves are for multimode fiber.

Figure 26. Attenuation vs. Wavelength [Ref. 34: p. 10]

to reduce delays. This method is least desirable since it adds heat, causes a shorter light source lifetime, and adds noise. [Ref. 35: p. 109]

The receiver or detector of an optical signal is a photodetector which generates a carrier replicating the source signal. These photodetectors are of two types, Avalanche Photo Detector (APD) and Positive-Intrinsic-Negative (PIN). Both detectors require amplification of the received signal before pulse detection circuitry can be used. This amplification adds noise into the receivers as well. Due to the low source power and high receiver noise, fiber optics systems make a tradeoff using the large bandwidth available to enable enough power to be received. PINs are used for small bandwidth or short haul systems due to their extremely linear response at low voltages. [Refs. 34: p. 22, 35: p. 130]

Another unique feature of fiber optics is the types of multiplexing it uses. Fiber cable is a natural medium to multiplex since the bandwidth of even small fibers exceeds the capacity requirements for most communications systems. Therefore, in addition to traditional TDM, fiber optic systems use Wavelength Division Multiplexing (WDM) in which different wavelengths of light are used as separate carriers much like the FDM divides its channels by frequency. Optical filters are used to combine and separate channels. [Ref. 34: p. 22-27]

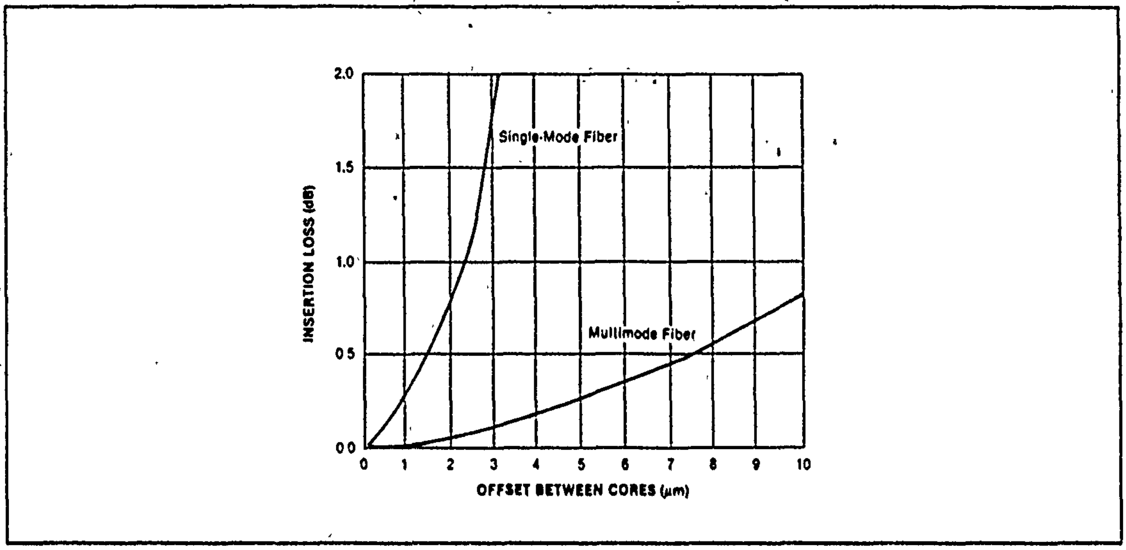


Figure 27. Insertion Loss in Fiber [Ref. 34: p. 14]

The only disadvantage of fiber optics today is the high cost of splicing due to the precision requirements. On long haul routes, however, fiber optics is now the most economical medium. As technology develops and splicing costs decline, short haul routes will soon be using fiber. The advantages of fiber optics are numerous. The bandwidth is extremely wide affording high capacity systems that are smaller and lighter. The non-inductive nature of fiber optics makes it immune to crosstalk and electromagnetic interference (EMI). Repeaters can be spaced further apart due to low attenuation losses. Fiber optics are virtually impervious to weather and moisture, and maintenance costs are dramatically lower than wire based systems. Fiber optics are safer. There are no high voltage lines, no sparks, and since optical fibers do not radiate energy, they are secure and hard to tap. Finally, the basic raw material used to make optical fiber is silica, and it is never likely to be in short supply. Without a doubt, fiber optics is the key to the future. Presently, fiber optics systems are in use that provide nearly 8000 voice channels on a single fiber the thickness of a human hair. [Refs. 36 : p. 272, 37: p. 64]

6. Microwave

Microwave links are used for both short haul and long haul situations and are especially good in situations where the route is over rough terrain, inaccessible terrain, or over water. Also, microwave systems are valuable in situations where rights-of-way will be expensive or difficult to obtain, quick installation is needed, or in areas with large anticipated growth. Microwave is a line-of-sight (LOS) system that travels between two microwave antennas normally mounted on buildings or mast structures built especially for them. The antennas are normally spaced 20 to 30 miles apart and can be retransmitted over a long route as many as 50 times, after which the signal runs into problems with distortion due to noise build up each time it is reamplified. The type of transmission can be either analog or digital. Analog transmissions are usually FM or SSB AM using FDM multiplexing. Receivers for analog systems are usually superheterodyne. The trend today is towards AM due to technology improvements in AM power amplifiers. Digital systems use QPSK or QAM with TDM multiplexing. [Refs. 25: p. 166-171, 22 : p. 4.1-4.21]

The frequency spectrum used in microwave systems is from 1.71 GHz to 40 GHz, which is roughly in the Super High Frequency (SHF) and lower Extremely High Frequency (EHF) range. Since high frequencies equates to short wavelengths, it is possible to use parabolic antennas that feature high gain for receivers and transmitters. As seen in Figure 28 on page 71, parabolic antennas allow the beam to be focused, typically to within one degree, making the system highly directional. Since these types of antennas have high gains, they are usually only 10 feet in diameter which allows them to be installed most anyplace. Also, the higher the frequency, the more focused the beam width can be which will allow even smaller antennas or lowered terrain clearances. However, antenna placement is critical due to the high frequencies used in microwave. The path between the two antennas must be clear of trees, buildings or other objects that may block or reflect the signal resulting in the transmission not being received or echoes being caused by the reflection. Other problems associated with antenna placement are diffraction and multipath. Diffraction is caused by insufficient terrain clearance that causes the signal to bend down toward the earth which causes the signal to be off target resulting in fading or signal loss. Multipath is where some of the signal is reflected and continues to the next microwave facility. When the two or more paths join at the receiver, they may be in phase, out of phase, or somewhere in between. If they are in phase, reception will be good, but fading will occur if the signals are received at different phases, and the signal will be lost if the paths are out of phase. [Ref. 22: p. 4.6-4.9]

Probably the most significant effects are from the atmosphere. Rain is the worst enemy of microwave, and effects are greatest at frequencies above 10 GHz. For this reason, long routes over rainy areas will use lower frequencies. Another technique used to defeat the effects of rain is the use of diversity where space diversity is the most common form. Sometimes just a slight change in the antenna height or transmission

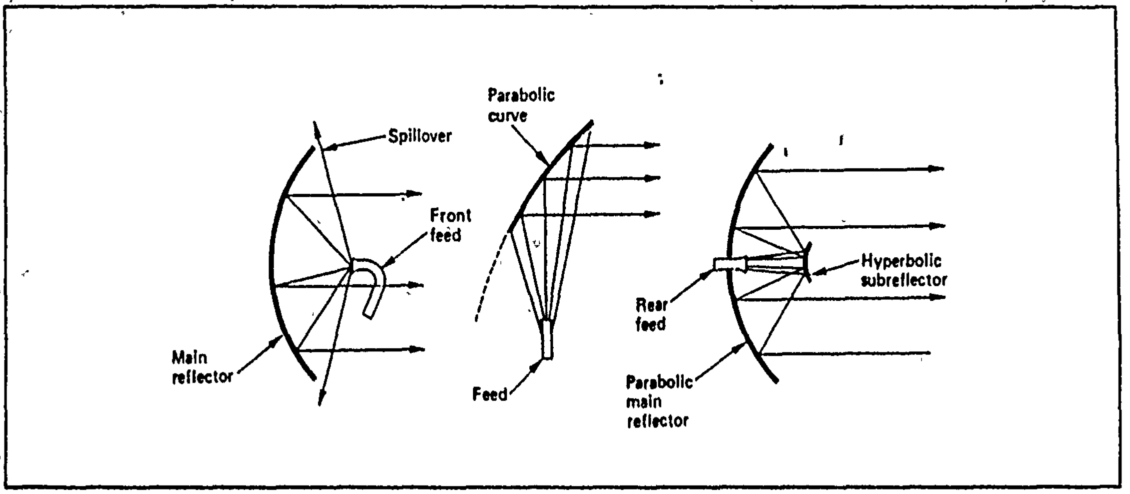


Figure 28. Various Parabolic Antennas Used in Microwave [Ref. 37: p. 71]

angle can be enough to overcome this problem. Other negative atmospheric effects are atmospheric multipath, ducting and subrefractive fading. All three of these are caused by abnormal temperature gradients that bend or channel the signal so that the receiver power is less than desired. In areas of heavy rain and adverse atmospherics, as much as 50 dB is added as a fade margin to ensure the percent probability of good signal reception. The last type of degradation is caused by interference. Unfortunately, microwave and many satellite systems share the same frequency spectrum. Also, in large cities, available bandwidth is limited. Therefore, the FCC tightly regulates frequency assignments to minimize interference by coordinating all systems within 125 miles of each other. [Ref. 27: p. 89-99]

7. Satellites

Communications satellites are simply a means to stretch radio frequency communications beyond LOS, and telephone communications satellites are essentially a microwave relay with a path length off 22,300 miles. The path length is due to the satellite being in geosynchronous orbit, meaning the position relative to a point on the earth's equator remains constant. To ensure the satellite is maintaining its station, thrusters using hydrazine as a propellant are fired every two to six weeks. The life expectancy of a satellite is not affected by the amount of propellant since satellites usually carry around 1300 pounds of hydrazine and only use 20 to 30 pounds per year. The service life is usually limited by the satellites' solar panels that supply the power. After seven years, the panels are expected to deteriorate to the point that the power output will be insufficient, and therefore this establishes the service life at seven years. However, most satellites have exceeded this life expectancy by a significant margin. [Ref. 38: p. 299-300]

The satellite system as a whole has a circuit reliability of 99.6% set by the International Telecommunication Union (ITU) which is expected to raise this standard to 99.8% soon. The satellite system consists of a control station, a network of earth stations, and a satellite. The control station is responsible for monitoring the satellite status and station keeping. Station keeping is critical since the equator is dotted with various satellites. The ITU and FCC are the agencies that approve geosynchronous orbital slots for communications satellites. Each slot is .1-.2 degrees in width and spaced every two to four degrees. The next element of the satellite system is the earth station which provides the up-links and down-links for the telephone signals. These signals are in two frequency bands: the C-band having a 5925 to 6725 MHz up-link and a 3700 to 4800 MHz down-link; and K_u-band which has a 14,000 to 14,500 MHz up-link and a 11,700 to 12,200 MHz down-link. The channels within these bands are 36 MHz wide and are spaced every 40 MHz. Since bandwidth is limited in the lower frequencies, frequency reuse is practiced by changing polarization of the transmission and using spot beams located on the satellite. Because the satellites are packed close together over the equator and frequencies are reused, the earth station must be able to tightly control its beam. In some cases, the antenna must be pointed with accuracy of less than one degree. On the other hand, since satellites share the same frequencies as microwave systems, the satellite spot beam must be controlled to two or three degrees. However, in some large metropolitan areas where a lot of microwave links are used, interference may still occur. This is why earth stations are located roughly 30 miles outside the urban area where the satellite down-link will not interfere. The signal is then sent via microwave into the city. [Ref. 22: p. 5.1-6.9]

The satellite itself can be one of two types, either a spinner or a three-axis stabilized satellite. Spinners are called that because the whole satellite spins to give it gyro-stabilization. The three-axis type achieves the same effect by using three internally mounted gyros, one in each plane of reference. Regardless of the type of satellite, each consists of a payload and a bus. The payload is the antennas, receivers, and transmitters. The receiver and transmitter are called a transponder, and each satellite normally carries 24, plus spare transponders and power supplies. The bus is the "housekeeper" in charge of command and telemetry subsystems, power management, and attitude control. [Ref. 38: p. 299-324]

Since satellites use the same frequencies as microwave systems, they are affected by the same atmospheric phenomena, especially rain. However, several additional considerations enter into satellite systems. First, the path length is 22,300 miles which, by comparison to microwave distances, is very long. Therefore, satellite systems rely on high equivalent isotropically radiated power (EIRP) for transmitters and high antenna gain. As a rule, the higher the power, the smaller the antenna. Earth station antennas range in size from 10 to 50 feet in diameter, but the most common size is 30 feet. Since satellites use parabolic antennas similar to microwave antennas, the beam width can be controlled. The diameter of the antenna determines the beam width and gain, and a tradeoff is usually made between beamwidth, EIRP, and antenna diameter. The relationship between beam width (θ), frequency (f), and antenna diameter (D) is given by the formula, $\theta = k\lambda/D$, where k is a constant determined by EIRP and receiver sensitivity. The other problem associated with atmospherics is look angles. If the satellite earth station has a low look angle, it must travel through more of the earth's atmosphere to reach the satellite and will have significantly higher path losses. Normally a satellite can "see" roughly one third of the earth's surface or up to 81.3° either side of the equator. Because low look angles cause significant loss, the practical limit is 5° less, or roughly 76°. Also, satellite antennas must not point into the sun which will cause noise in the system and make the signal unreadable. [Ref. 34: p. 49-76]

In addition to frequency reuse, satellites use Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA) to make more efficient use of bandwidth. FDMA is when several SSB carrier telephone channels are multiplexed onto one baseband carrier transmitted by the earth station simultaneously. The satellite adjusts the passband within a multi-carrier transponder to receive each channel at the required capacity. This scheme maximizes use of the frequency allocation but runs into crosstalk problems at high power levels. However, the trend is toward TDMA because of its efficiency and adaptability to all digital systems. In TDMA, each user accesses the satellite using the same carrier frequency but at different time slots using very short burst transmissions. The time slot is synchronized by a master station that sends a sync pulse to all earth stations. During any given time interval, the up-link is shared with all other stations on the network. The master station defines a superframe by designating transmit slots for each earth station and may change these slots to accommodate changing system requirements. The most sophisticated type of TDMA is when these slots are allocated based on instantaneous demand. This type of system is called Demand Assigned Multiple Access (DAMA) and is very expensive, but it is the most efficient technique available. [Ref. 34: p. 72-75] Another feature of TDMA is the ability to increase power that can overcome the effects of rainstorms. This is due to the burst nature of TDMA and the fact it uses the entire bandwidth. The FDMA technique of only using a small portion of the bandwidth requires the available power to be subdivided among the various frequency bands which causes a lower power level per channel. [Ref. 15: p. 110]

The first telephone communication satellite was put in service in 1965 by Intelsat. Since then, Intelsat has its sixth generation in service providing over two thirds of the world's satellite telephony. Other major telephone satellites include Telstar (AT&T) and Spacenet (GTE). However, satellite links for domestic telephone routes are not widely used due to the inherent delay in the signal. It takes .24 to .275 seconds for a signal to reach the satellite depending on look angle. Since the telephone uses 85% voice traffic, this means for a speaker to get a response to a spoken statement would be twice this time, or roughly .5 seconds. In voice communications, this delay is considered to be poor quality in comparison to cable systems. Considering this confusing delay when using satellites and advances in fiber optics, the use of satellites by domestic long distance traffic will probably decrease. At the present state of technology, satellites are not cost effective for distances under 1500 miles. However, satellite costs are independent of distance and are used primarily for overseas telephone links. [Refs. 22, 25]

8. Submarine Cable

The first transatlantic cable was put into service in 1858, but was only capable of telegraph traffic. Due to the complexities of transmitting voice underwater, the first true telephone cable was not laid until 1950 between Florida and Cuba, but it is still in service today. Then in 1956, the first Transatlantic Telephone cable (TAT-1) went into operation between Europe and the U.S. Since then, hundreds of submarine cables have been laid worldwide using coaxial cable as the medium. Finally, in 1988 the TAT-8 transatlantic cable was laid and bec...ne the first major fiber optic submarine cable. [Refs. 39: p. 14, 40: p. 1-23]

Submarine cable systems must meet more rigid design criteria since the environment is generally more harsh and maintenance is more difficult. The system is essentially the same as a terrestrial cable system, whether it be coaxial or fiber, but the system must be able to: withstand depths up to three miles; provide a pressure and water-proof environment for the wires or fiber; provide a low resistance insulated power feed wire for the repeaters; and, in the case of fiber cable, ensure minimum axial stress is placed on the fibers. This latter requirement is due to the unique feature of glass fibers. When put under tensile stress and exposed to moisture, the fibers will deteriorate.

Therefore, by ensuring that water doesn't penetrate the casing and by relieving stress, the fibers will last indefinitely, well beyond a submarine cable's 25 year design life expectancy. [Ref. 40: p. 251]

Cables can be manufactured that float near the bottom, or more commonly lay on the bottom. Submarine cables in fishing areas or near coastlines are buried using a device called a seaplow. To protect the cable, an outer jacket of polyethylene is used and fibers are coated using teflon or a polyester elastomer. Since underwater bottom contours are not flat, it is important to manufacture the cable with strengthening members to make it more rigid to keep radii of curvature to acceptable levels. Any curvature exceeding the maximum will result in signal loss. Depending on the type of cable, repeaters are used to ensure signal loss is kept within tolerances. The TAT-8 system spaced its repeaters every 25 miles, whereas some older systems required repeaters every six miles. As technology expands, the distance between repeaters will grow. Already a repeaterless system spanning 100 miles is in use. The power for the repeaters comes from a wire usually located at the core in the submarine cable and provides approximately 12,000 volts DC. The power is fed from both terminal ends and uses the sea water as a return to complete the circuit. Voltage is high to ensure ample power is available to the center repeaters. System monitors control power levels to prevent power surges from damaging equipment located closer to the terminal end. [Refs. 39: p. 106, 35: p. 40-43]

The biggest threat to submarine cable is from satellite telephony. Over the years, submarine cable systems have been masters at compressing more channels into a limited bandwidth system. Two techniques unique only to submarine cable are the use of TASI and use of 3 KHz channels vice the normal 4 KHz. The combined effect of these two techniques is nearly a five fold increase in channel capacity without significant voice degradation. Although costs of submarine cable telephony are higher than satellite costs, the use of bandwidth saving schemes and future growth using fiber optics is bringing the costs to where submarine cable is competitive again. However, despite the cable's disadvantages, it provides the same quality of service at a 99.7% or better reliability factor and provides diversity in transoceanic telephony. As noted in an earlier section, satellites are hampered by a .5 second delay, thus making a switched satellite network too confusing for voice traffic. On the other hand, submarine cable experiences an insignificant delay of .030 seconds on a transatlantic call and can be switched over several systems if needed. Additionally, cable is not affected by the atmosphere, interference or rain. However, the most important aspect of submarine cable is that there is no limit on system expansion. Conversely, satellite orbital slots are being used up, as is the radio frequency (RF) spectrum. [Ref. 39: p. 106-178]

9. Digital Media

The emphasis in the past decade has definitely been toward digital service with the end goal of an all digital end-to-end telephone system. Therefore, a short section will be devoted to Digital Service (DS) standards. The basic digital service is DS-0 which provides a capacity of 64Kbps. Next is DS-1 at 1.544 Mbps, followed by DS-1C which provides 3.152 Mbps. All three of these can be transmitted over any medium. The larger capacity circuits are: DS-2 at 6.312 Mbps; DS-3 at 44.736 Mbps; and DS-4 at 274.176 Mbps. These circuits use microwave or fiber optics as the medium.

The most common type of digital service is called T-1. T-1 refers to any digital transmission using the DS-1 data rate. It consists of 24 voice or data channels at 64 Kbps per channel plus 8 Kbps of overhead for synchronization for a total of 1.544 Mbps. T-1 has become very popular as a leased line that can provide a large capacity for minimal costs for many large users. T-1 systems use M24 and M44 voice channel banks or bandwidth managers to combine voice and data.

10. Cellular Telephone

The idea of cellular telephone is an outgrowth of the mobile radio communications systems that became popular in the 1920s on board ships and aircraft. During the 1930s, these systems were still in moderate operational use and mostly experimental. It was not until the end of WWII that the frequency spectrum was opened up, and that allowed the idea of mobile radio telephone to take off. Mobile radio telephone had several advantages in that it could communicate over long distances, but it was hampered by the need for high power output, low frequencies, and large antennas. Therefore, the concept of a cellular system using low power levels over a small geographic area provided the solution to overcome these handicaps. Non-cellular radio telephone got its start in 1946 when a system was constructed in St. Louis. The system required special procedures and operator intervention which caused the Improved Mobile Telephone System (IMTS) to be built in 1960. IMTS allowed subscribers to call just like on a regular telephone. [Refs. 34: p. 248, 22: p. 22.3]

As technology improved and the idea of cellular phones continued to grow, the issue of cellular telephone service became stalled due to frequency spectrum allocation issues. Attempts to gain an adequate frequency allocation were attempted by Bell Telephone in 1947, 1949, and 1958, but were denied by the FCC. In 1968, the Bell system had proven the cellular concept with a 10 cell system used for the railroad's metroliner service between New York and Washington. In 1974, this system helped convince the FCC to reallocate 40 MHz of spectrum for public mobile telephone service to be used by wire line common carriers and 30 MHz for private mobile radio use. As the first non-experimental cellular network, Illinois Bell installed the Advanced Mobile Phone Service (AMPS) in Chicago in 1978 which was followed in 1979 by a similar system in Washington, D.C., installed by Motorola. The systems proved to be popular and started a rapid shift in attention to cellular phones. As interest from the private sector grew, the FCC decided to promote competition by splitting the 40 MHz allocation in two, 20 MHz each for wire line common carriers (local telephone company) and 20 MHz for the non-wire line common carriers or radio common carriers (RCCs). This was discussed under regulation in chapter two. The actual frequency allocations for non-wire line common carriers (RCC) are 825-835 MHz for mobile unit transmission and 870-880 MHz for cell site transmission. For wire line common carriers (telephone companies), the mobile unit transmission range is 835-845 MHz and the cell site transmission range is 880-890 Mhz. [Ref. 22: p. 22.4]

The objective of the cellular telephone system is to provide efficient use of bandwidth available in order to provide nationwide service to the maximum number of subscribers at a quality of service comparable to regular telephone standards. Presently, the quality of service needs to be improved as only 90% of calls are considered good to excellent, 9% are rated fair and 1% are rated poor or unusable. [Ref. 34: p. 263]

The design of the cellular telephone centers around the cell and the frequency reuse plan. In Figure 29 on page 78, it can be seen that the shape of a cell was chosen to be hexagonal so that areas could approximate a circle without any gaps in coverage. Circular cells would be the best shape to represent the theoretical coverage, but circles do not fit together without having to be overlapped to avoid gaps in coverage. On the other hand, rectangles or squares would fit neatly together, but do not represent the theoretical coverage of a radio transmitter. Therefore, hexagonal cells were chosen as a compromise between a circle and a square. Note that in the figure, the same seven cells or dirichlet regions form a cluster and are reused in the same pattern. Although cells are designed as hexagons, actual conditions, especially land use permits, may dictate another shape of the cell due to transmission characteristics. Cell sizes vary from several thousand feet in diameter to several miles, dependent upon terrain, population, regulation, environmental conditions, and power limitations. The effects of these factors can be seen in Figure 30 on page 79 where the left side depicts the ideal cell coverage with the transmitter in the center of the cell, and the actual coverage attained is shown on the right hand side. Most of the area not covered is a result of shadowing caused by signals being blocked by hills, buildings, and foliage. Because of results like this, site surveys as well as overall system design must be carefully analyzed to provide the optimum coverage. [Ref. 22: p. 22.6]

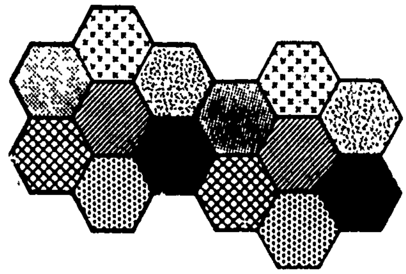


Figure 29. Frequency Reuse Patterns [Ref. 22: p. 22.1]

There are two nearly identical cellular systems within each cell operated by the local telephone company and the RCC. Each system is comprised of a cell site and/or base station and a Mobile Telephone Switching Office (MTSO). The cell site is the remote transmitter and receiver that is linked to the MTSO via land line. The cell site has a transmitting power of 25-35 watts which is just enough to cover its entire geographic cell area but not interfere with adjacent cells of the same frequency. Cellular systems also use directional antennas to minimize interference. These antennas take advantage of the cell's shape and use 180°, 120°, or 60° coverage. The positioning of the cell site is usually in the center of the cell, but cells may have several cell sites. The number and location of cell sites is dependent upon EIRP, receiver sensitivity, environmental factors, and system design reliability. In Figure 31 on page 80, a cell site using 35 watts EIRP and an antenna height of 100 feet can only reach a little over five miles. Note that the graph is logarithmic, and an order of magnitude increase in EIRP does not even double the range, whereas a modest increase in antenna height produces much better results. This is because EIRP follows the inverse square law for an isotropically radiated antenna which is given by the formula, $P_R = (P_T\lambda^2)/(4\pi r)^2$, where P_R is the received power, P_T is the transmitted power, r is the range and λ is the wavelength of the frequency in use. Since $\lambda = C/f$, where C is the speed of light (3.0×10^8 m/sec), and λ for the frequencies 825-890 Mhz will be a wavelength less than one, both λ and r will have the effect of reducing P_R. The last component of the cellular system is the MTSO. The MTSO is the

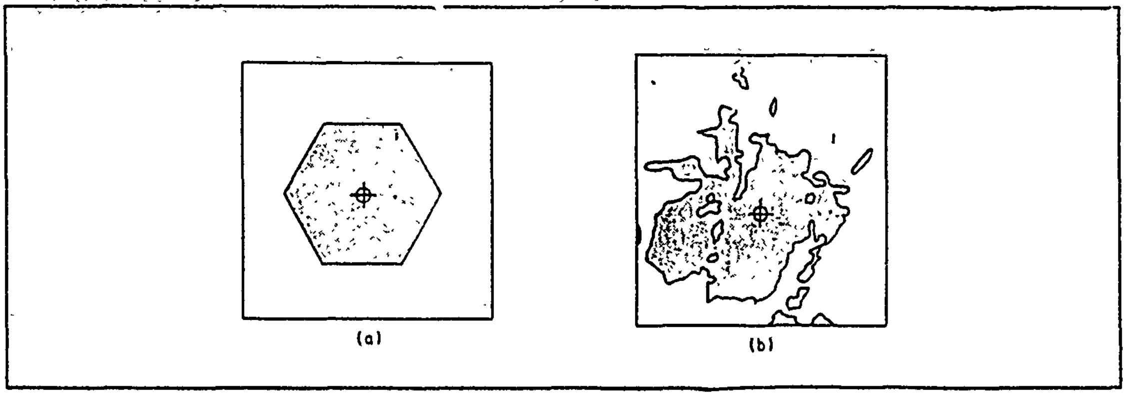


Figure 30. **Ideal Cell Coverage vs. Actual Cell Coverage** [Ref. 22: p. 22.6]

switching center for several clusters of cells and is tied directly to the LEC's CO via a land/line. The relationship of the components is depicted in Figure 32 on page 81. [Ref. 34: p. 250]

All cellular systems operate with 333 channels, each 30 MHz wide that must be used for all cells in the cluster. Since out-band signaling is used in 21 of these channels, only 312 channels are left for voice traffic. Since a portion of these 312 channels must be allocated to each cell, the general idea is to keep the number of cells (N) in a cluster as small as practical. The limitation on this becomes a tradeoff between coverage, power, and interference from adjacent clusters. The tradeoff usually results in N values of four or seven. Therefore the number of channels per cell is given by A/N, where A is the total allocation (312). For a system using N=7, each cell will have 44 or 45 channels, whereas a system using N=4 will have 78 channels per cell. In areas of high traffic, channel borrowing may occur to increase the available channels for that cell. [Refs. 34: p. 268, 22 : p. 22.9]

The system is limited by factors other than limited spectrum, most noticeably design considerations due to the mobility requirement (weight, power, size) and the fact that cellular phones rely on RF propagation. The mobility aspect limits the antenna size of the mobile unit (normally less than 5 feet) and mobile transmitter power is limited to between .6 and 3 watts. The output power of the mobile unit is controlled by a computer at the cell site to avoid interference. Antennas at the cell sites are typically four to five feet long and mounted 100-300 feet high in suburban areas and over 300 feet in urban areas. Since the signal is being passed to and from a moving unit, the problem of multipath represents an interesting problem. Earlier in the chapter, multipath was

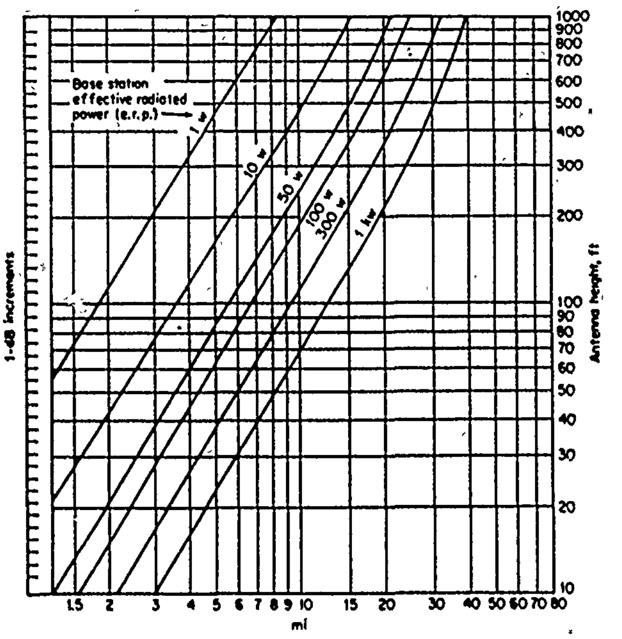


Figure 31. Base to Mobile Unit Range

defined to be the loss, gain, or fading of a signal due to the signal being received from multiple paths caused by reflections. In cellular telephone, if the straight line path is blocked, multipath will usually occur. Typical results of multipath are seen in Figure 33 on page 82 where in a period of only 1.8 seconds, and over a distance traveled by the mobile unit of only 38 feet, the signal varied by 40 dB. This all too often results in a severely degraded signal. To combat this, two types of space diversity are used called microscopic and macroscopic diversity. Microscopic diversity can be seen in Figure 34 on page 83 where the cell site antennas are located in pairs within a wavelength of each other using both horizontal and vertical diversity. Macroscopic diversity is seen in Figure 35 on page 84 where the cell site uses numerous transmitters to obtain direct path contact. Macroscopic diversity also assists in maintaining better coverage due to other limitations as previously discussed. The other consideration is RF propagation which is affected by external noise due to man-made emissions such as power lines and car ignition systems as well as natural phenomenon like galactic noise and lightning all of which have an adverse effect on cellular phones. Again the answer in overcoming these problems lies in system design tradeoffs mentioned earlier. [Refs. 34 : p. 252, 22: p. 22.15].

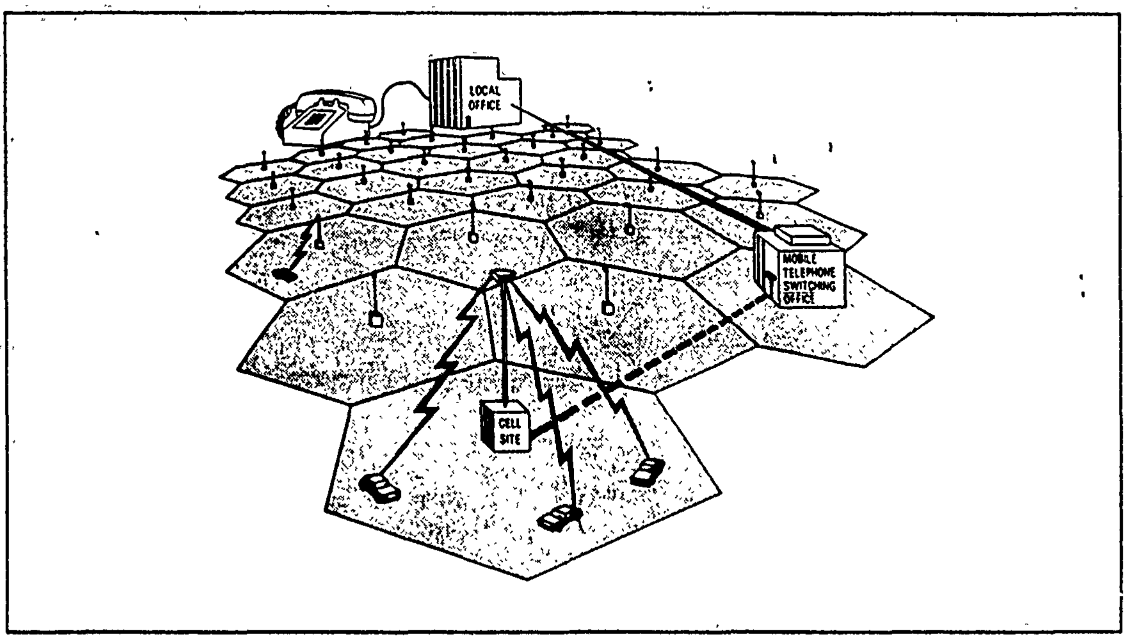


Figure 32. Components of a Cellular Phone System [Ref. 22: p. 22.3]

The cellular telephone system works by requiring all cellular terminals to be able to tune to any frequency immediately and by designing the ability of the cell site or base station to keep track of mobile units and hand them off when the cell boundary is crossed. In order to receive an incoming call, the receiver in the mobile unit must be turned on which allows the receiver to scan all control channels for the system. As the receiver scans the control channels, it marks the strongest channel for use and then monitors this channel for incoming calls. Incoming calls will be detected by the receiver picking out the mobile unit's telephone number from the digitally modulated (FSK) stream being transmitted on all control channels by the system's cell sites. The mobile unit's number is just like a normal 10 digit telephone number. The first three digits are the area code and are used only if the incoming call is being made from a different home area. A home area is made up of one or more area codes that determine the local cellular telephone calling area. Calls made from outside the local calling area will pay additional charges based on the use of an IXC carrier needed to connect the two home areas. Calls made outside of the home area are called roaming calls. Conversely, a call being made by the mobile unit to a telephone number with the same home area code needs only the seven digit portion of the number. Calls made to area codes outside the mobile unit's area code must include the three digit area code. For example, if a mobile unit's telephone area code is for Atlanta and the mobile unit is located in Minneapolis,

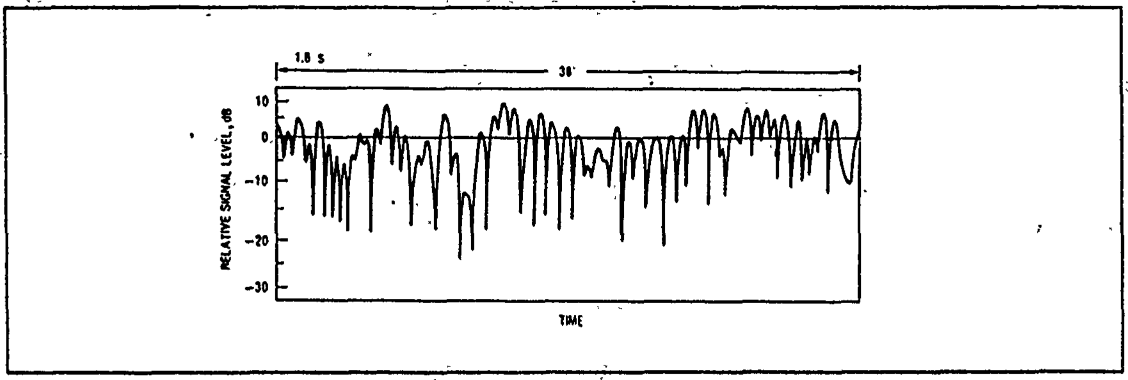


Figure 33. Typical Received Signal Variations Due to Multipath [Ref. 22: p. 22.15]

the user only needs to dial the seven digits to call Atlanta. This is because the system takes into consideration the user does not know when he is crossing boundaries for the telephone area codes, so the system always assumes numbers dialed without an area code are for the home area. Once the calling party has dialed the number, the number travels through the PTN which routes it to a central computer in the mobile user's system. The mobile user's system cell sites then broadcast the number over control channels for all cells. This is because the cell site does not know where the mobile unit is until the site starts to track the user after a call is established. Once the mobile unit detects its number, it transmits its identification back to the cell site. The cell site signals the mobile unit via the control channel which channel to use for the call. At this time, the mobile unit will automatically tune to this channel and alert the user to the incoming call. The incoming call can be from another mobile unit or from a normal telephone. [Refs. 34: p. 263-272, 22: p. 22.2]

A mobile unit is assigned a channel number that corresponds to a particular cell, but because of differences in propagation patterns, the user may be tuned to a channel of one cell while physically located in another cell. This raises the question of how calls are transferred between cells as the mobile unit crosses cell boundaries. This process is called a hand off and generally occurs at cell boundaries. Since the objective is to provide the best S/N ratio, the cell site determines when a call hand off should occur based on absolute signal strength for the assigned channel in comparison to neighboring cells. This is accomplished by the cell site measuring the signal strength every five to ten seconds and comparing it to a predetermined threshold value for that cell site. If the threshold is achieved, no hand off is required. However, if the absolute signal strength is below threshold, then a hand off may be required. To see if a hand off is required, the

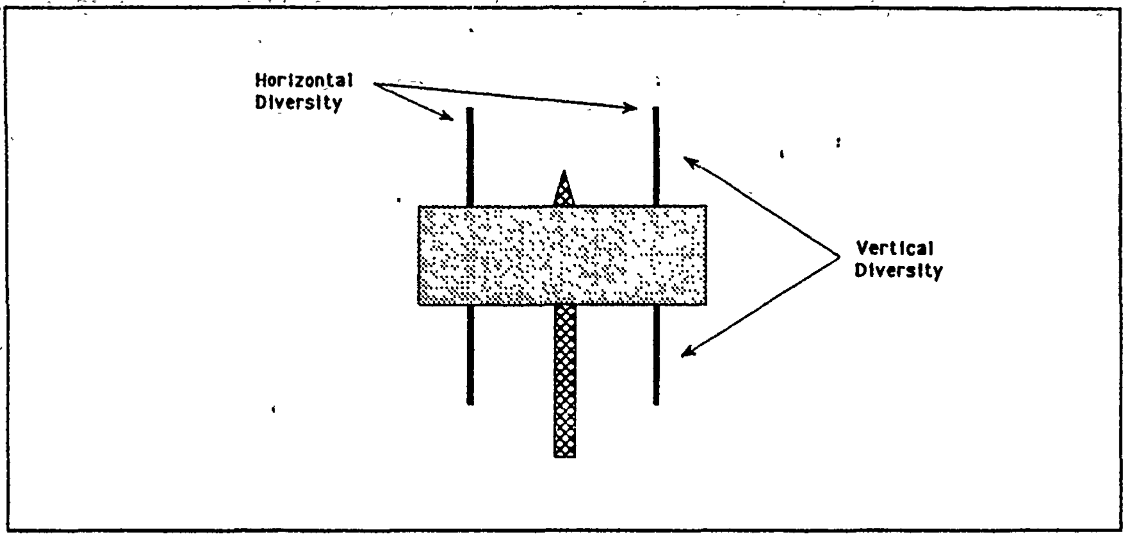


Figure 34. Microscopic Diversity

cell site requests the MTSO to collect neighboring cell sites' signal strengths so a comparison can be made. The MTSO collects the signal strengths and passes the information to the requesting cell site. If a neighboring cell site has a significantly better signal strength, the requesting cell site will ask the MTSO for a hand off. The MTSO searches its memory for an available channel for the new cell site and passes the assignment to the mobile user via the requesting cell site and to the new cell site directly. Both the mobile user and new cell site automatically tune to this channel, and the call proceeds as usual. Each hand off requires a digitally transmitted message that contains the new channel number to interrupt the call. However, the interruption is brief and is not noticeable to the users on either end. [Ref. 34: p. 273-274]

There are two types of channel supervision that are needed for cellular phones. Recall that supervision is essentially the process of detecting changes in hook status. The first type is Supervisory Audio Tone (SAT) which is a continuous out-band tone generated by the cell sites and modulated into every voice channel. The tone is automatically retransmitted back to the cell site by a transponder in the mobile equipment. The cell site receives the returned tone that indicates a call is in progress. If the tone is lost for a period of approximately five to ten seconds, the cell site will assume the call was terminated and will release the call. SATs are similar to the 2600 Hz signals used in the PTN, except that the frequencies used in cellular phones are 5970, 6000, and 6030 Hz. Only one of these frequencies will be transmitted at a time. The second type is the

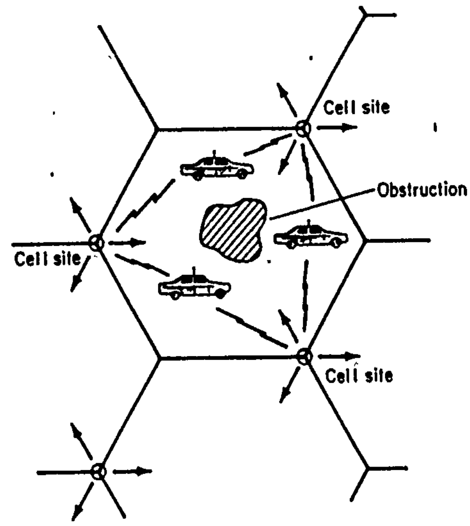


Figure 35. **Macroscopic Diversity**

signaling tone (ST) which is a short burst at 10 KHz and is present when alerting the user of incoming calls, being handed off, or disconnecting. [Ref. 22: p. 22.12]

One problem associated with limited control channels is the possibility of seizure. Since all mobile units use the same control channels to set up calls, collisions can occur. To avoid this, transmissions from cell sites to mobile units include one bit in every digital frame dedicated to call collision avoidance. This bit is called the "busy/idle" bit and is set to "busy" when the cell site detects a collision occurring. This bit is received by the mobile units which automatically wait a random time before retransmitting. The process is similar to carrier sense multiple access which is a technique used in computer networks to avoid seizure. [Ref. 22: p.22.12]

V. INTEGRATED SERVICES DIGITAL NETWORK

A. OVERVIEW

Integrated Services Digital Network (ISDN) is an evolutionary worldwide public telecommunications network that the PTN is moving toward. The concept is simply an end-to-end digital network that will replace the mixed analog and digital telephony systems of today. The present PTN was built as two separate entities, transmission and switching, or in telephone jargon, the outside plant and the inside plant, respectively. Consequently, these two entities often required unnecessary conversion from one form to the other for proper switching and transmission.

In an analog system, each switch must multiplex the outgoing signal in order to transmit and demultiplex the incoming signal to receive it. This process must be done at every switching center. It adds expensive FDM channel banks and builds unwanted noise in the signal as a result. This process can be seen in Figure 36. By designing and building an all digital system, the expensive channel banks are eliminated, and the signal only needs to be multiplexed once as seen in Figure 37 on page 86.

The term integrated in the name ISDN refers to the integration of voice, data, and video on one network as well as the integration of the transmission and switching systems into an all digital system. This concept will result in economies of scale, simplified telecommunications structure for most users, and less maintenance. [Ref. 19: p. 585-589]

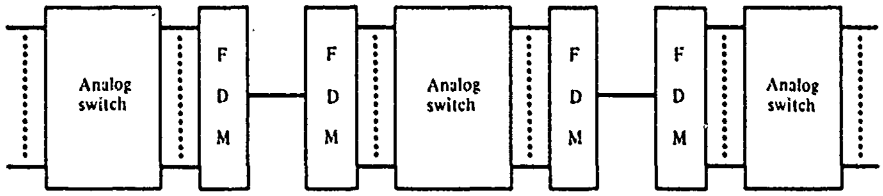


Figure 36. Frequency Division Multiplexing in an Analog System [Ref. 19: p. 586]

Since a great amount of capital has been sunk into copper lines all over the world, ISDN will evolve from present telephone and other systems to capitalize on this invest-

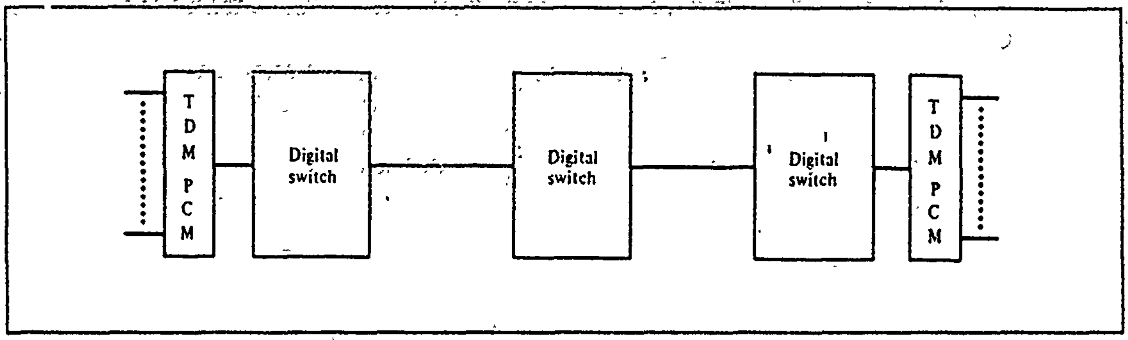


Figure 37. Time Division Multiplexing in a Digital System [Ref. 19: p. 586]

ment and to enable the maximum number of users. This evolutionary concept supports the key objectives of ISDN that are outlined below [Ref. 19: p. 589]:

- Standardization of protocols and standards to permit the maximum number of users worldwide
- Protocols for services provided will be transparent to the user
- Competitive functions separated as much as possible. In some countries services such as videotex and electronic mail will be separated in order to allow them to be competitively offered
- Ability to provide leased and switched services
- Ability to keep tariffs cost related and independent of data type
- Smooth migration from existing plants to an all digital system
- Ability to multiplex channels to provide small user demands as well as large user demands

B. STANDARDS AND PROTOCOLS

There are many organizations involved in developing ISDN standards, but the International Telegraph and Telephone Consultive Committee (CCITT) is the predominant organization. In 1984, the CCITT Red Book laid out the basic standards for ISDN. These standards are still maturing and will allow circuit switched, packet switched and non-switched (leased) connections. [Ref. 19: p. 595]

The CCITT standards (actually recommendations) are based upon the International Standards Organization (ISO) Open System Interconnection (OSI) reference model that defines seven layers that standardize computer communication development so that equipment built by different countries or companies can communicate with each other. However, the OSI reference model does not specifically address the unique aspects of ISDN. Some of these unique features are:

- Multiple related protocols where a D channel is used to set up, maintain, and terminate a B channel signal (channels will be discussed in the next section)
- Multi-media protocols required to handle voice, data, and video simultaneously
- Multi-point connections allowed by conference calling. [Ref. 19: p. 606]

C. ARCHITECTURE

The actual architecture of ISDN is quite simple. As seen in Figure 38, the CPE, Local Area Network (LAN), PBX or other equipment is connected directly to an ISDN interface. This interface will be described later. The interface is connected by a "digital pipe" sized to meet the customer's needs. Obviously, a residential customer does not need the capacity of a large corporation. This pipe connects to the ISDN CO which then routes the call over the proper portion of the network. The transmission structures of ISDN centers around three types of channels [Ref. 22: p. 14.35]:

- B channel - 64 kbps - circuit or packet switched
- D channel - 16 or 64 Kbps - packet switched only
- H channel - 384, 1536, or 1920 Kbps - circuit or packet switched

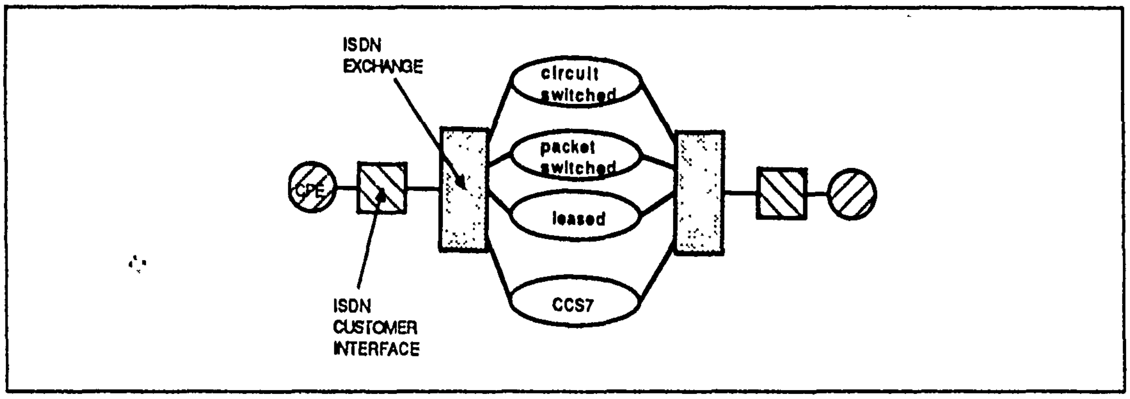


Figure 38. ISDN Architecture [Ref. 19: p. 587]

The B channel is the basic channel used to carry the actual information of the call. There are three kinds of connections that can be made over a B channel: circuit switched, packet switched, and semi-permanent (leased). The B channel rate was chosen to meet the minimum acceptable data rate that could adequately reproduce voice digitally and still be able to use existing equipment and lines. However, technology has continued to lessen the requirement, as 32 Kbps can now reproduce voice with the same quality. B channels are used for video, high quality audio and other high speed data needs. [Ref. 22]

The D channel serves two purposes. First, its primary function is to carry the necessary control information for circuit and packet switched calls. Secondly, it can provide a source for low data rate transmission when not required for signaling. H channels are essentially the same as B channels and are used for high data rate applications. [Ref. 22]

As noted previously, the common thread in ISDN is the 64 Kbps basic rate of the B channel. In order to provide the wide variety of services, two transmission structures will be used. First is the Basic Rate Interface (BRI) which will use two 64 Kbps B channels plus one 16 Kbps D channel. Since framing, synchronization and other overhead require 48 Kbps, the total bit rate required to support the BRI is 192 Kbps. The BRI is often referred to as the 2B + D interface. The second structure is the Primary Rate Interface (PRI) which is 23 64 Kbps B channels, one 64 Kbps D channel, plus 8 Kbps of overhead for a total of 1.544 Mbps. This structure was chosen for use in the U.S., Canada, and Japan to match the T-1 data rate. In Europe, the structure will be 30 B channels, plus a D channel and overhead that increases the rate to 2.048 Mbps. Note that both the BRI and PRI structures use a D channel to signal. It is important to note that ISDN will use CCS, specifically CCS7, as the signaling system for circuit switched applications whereas CCITT protocols X.25 and I.451 will apply to packet switching applications. [Ref. 19: p. 597-600]

The evolution to ISDN can be described as a two phased approach that will proceed from analog to digital in phase one and copper to fiber optics in phase two. Nearly all new equipment being installed today is digital with cabling being optical fiber. As it becomes cost effective to replace old analog equipment and copper cables, ISDN will move closer to reality. Although there are several locations where ISDN is already installed, ISDN is not expected to be fully implemented until the late 1990s. [Ref. 19: p. 585-587]

There is considerable debate over the merits of ISDN as many people in the telecommunications business feel standards lock them into obsolete technology. Additionally, some feel the data rates offered are not sufficient to meet the demands of tomorrow. For example, High Definition Television (HDTV) requires a minimum of 92 Mbps. The best ISDN can offer is 1.92 Mbps, far short of the requirement. Therefore, another form of ISDN has been proposed called Broadband ISDN (B-ISDN). The data rates associated with B-ISDN will range from 30 to 140 Mbps per channel. The number of channels will depend on the user needs, but the standard interface has been set at 600 Mbps to the user and 150 Mbps from the user. The difference in capabilities is because, as a rule, the user will receive more than send. For instance, a user will only want to receive HDTV, not send it. [Refs. 22, 20: p. 5.1-5.4]

Finally, as alluded to earlier, there are several ISDN customer interfaces that can be used depending on the user's equipment. Figure 39 depicts the various "black boxes" an ISDN user-network interface will have. The device labeled NT1 is the Network Termination 1. It provides only the physical and electrical connections in concert with the OSI reference model. An NT1 is mandatory for all ISDN hook ups. An optional Network Terminator is the NT2, which must be used in conjunction with the NT1. The NT2 provides switching, routing and other functions. Examples of NT2s are PBXs and LANs. If the two NTs are combined, they are called an NT12. Furthermore, customer Terminal Equipment (TE) is broken down into two categories. First, TE1 is ready to use digital equipment such as a digital telephone. TE1 equipment plugs directly into the NT1, NT2, or NT12 as desired. The second type is the TE2 which is non-ISDN compatible and requires a Terminal Adaptor (TA).]Ref. 19: p. 601-606]

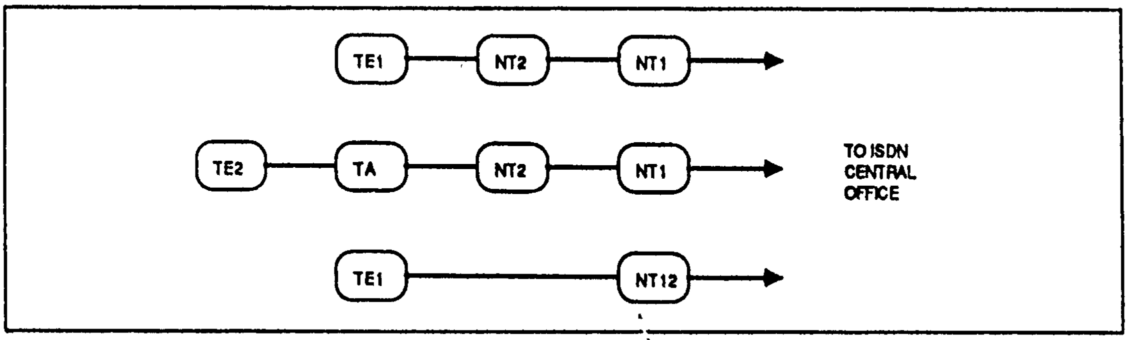


Figure 39. Various Types of User-Network Interfaces [Ref. 19: p. 603]

In conclusion to this chapter, it is appropriate to mention a few words concerning the implementation of ISDN. Certainly ISDN has many advantages, but there are those in the telecommunications field that believe that ISDN is already obsolete. Their belief stems from the fact that many of today's digital applications require higher data rates than those being offered by a single B channel on ISDN. Some of these applications are high resolution video screens such as Video Graphics Adaptor (VGA) screens, full motion video, and networking with Local Area Networks (LANs). The data rate of the B channel would cause considerable bottlenecks for high speed LANs and the refresh rate of the screen would be too slow for the remaining applications. However, these critics are overlooking the capabilities of ISDN H channels and the future implementation of B-ISDN which will satisfy almost every need. Furthermore, at some point in time in the design of a system a set of standards must be locked into place so that all the participants will know what to build. For those needing even more capabilities, the solution is to build their own private network. The concept of ISDN is the logical follow on to the original concept of universal telephone service. ISDN will provide anyone connected to the PTN the ability to use its services. Consequently, the move towards ISDN continues.

VI. THE MILITARY TELEPHONE SYSTEM

A. INTRODUCTION

Today's military telephone system is a result of numerous regulatory and technological changes that occurred in the U.S. starting in the late 1970s and culminated in the breakup of AT&T. As a result, the government lost its bulk rate discount tariff called TELPAK, and telephone costs increased dramatically. Also, changes in policy, user requirements, and sophistication of espionage methods caused DOD to reexamine the role of the telephone in communications. The result of this analysis was the concept and implementation of the Defense Switched Network (DSN). Primarily due to technological changes, it was found that many telephone services were cheaper to lease than to buy. Therefore, not surprisingly, the architecture of the Military Telephone System (MTS) is almost identical to the civilian architecture since the MTS uses mostly commercial equipment today. The differences arise mainly due to the military requirements for security, reliability, and survivability. Nearly every phase of the MTS is under an upgrade program of some sort that is affected by the Defense Switched Network or will be incorporated as part of it. Therefore, discussion of the DSN will constitute the majority of this chapter.

B. DEFENSE SWITCHED NETWORK

1. Overview

The Defense Switched Network is the command and control telecommunications network for the U. S. Armed Forces and Department of Defense (DOD). The DSN is an evolutionary program that will consolidate the best commercially available components and other specifically designed military components into an end-to-end digital network capable of handling voice, data, and video. The program stresses the use of commercially available hardware to avoid delays and costs associated with the research and development of a militarized system and to capitalize on the ability to advance capabilities as the commercial upgrades occur. Therefore, the majority of funding for DSN is Operations and Maintenance (O&M) which funds the leasing of transmission and switching equipment from commercial sources. Funding for the DSN will range from $552 million in FY90 to $703 million in FY95 with the majority being funded by the Navy and Air Force. Approximately 75% of each year's DSN budget is O&M. [Ref. 41: p. ES-20-22]

DSN will encompass a worldwide upgrade that will achieve effective military command and control, not only to transmit battle orders, but to coordinate personnel movements, order parts, or perform other administrative matters. Most importantly and according to the Defense Communications Agency (DCA), it is considered the primary communications system for national crisis, nuclear war, post attack, conventional warfare, and peacetime operations. There will be three levels of DSN users: the special command and control user; the command and control user; and the general user. [Ref. 41: p. ES-1-9]

The special command and control user has direct, unrestricted access to the DSN because of operations, planning, or controlling responsibilities through all phases of national crisis, alerts, conflicts, and war. This user is identified by a validation process controlled by JCS, the CINC, the military agency, or DOD. The special command and control user has flash override or flash precedence. Those key elements with flash override are limited to the President, Secretary of Defense, Joint Chiefs of Staff, and unified and specified commanders-in-chief declaring DEFCON ONE or defense emergency. Flash precedence is more widely distributed and is reserved for calls pertaining to command and control of military forces essential to defense and retaliation, critical intelligence essential to national survival, diplomatic negotiations leading to arresting or limiting hostilities, dissemination of civil alert and reporting catastrophic events of national or international significance. The command and control user is generally the operations oriented user who does not meet the requirements of the special user, but still needs precedence over non-operational users. The command and control user is limited to immediate or lower precedence calls. Finally, the general user has the requirement to use DSN, but does not meet the requirement for any urgency and has access only through routine precedence.

The benefits of DSN are increased survivability, responsiveness, security, cost effectiveness, and interoperability. As for survivability, the number of DSN nodes will increase between four to 20 times the present number of AUTOVON nodes depending on location, and the nodes will be more widely distributed as well. They will be interconnected by a variety of transmission media and sophisticated routing. End Offices (EO) will be dual-homed and will be interoperable with North Atlantic Treaty Organization (NATO), Korean, Japanese and all U.S. private telephone systems. This will increase the number of nodes and paths that increase the survivability. DSN will have improved routing strategies using commercial as well as dedicated government systems.

Also, the capability of rapid reconfiguration using mixed media and locating switches outside nuclear target areas increases survivability. [Ref. 41: p. ES6-ES9]

Responsiveness will be increased due to the increase in nodes, paths and ability of the system to find more efficient trunking and switching. The delays of today's Multi-Level Precedence/Preemption (MLPP) will be a thing of the past. MLPP will still be available if absolutely necessary, however, Precedence Access Thresholding (PAT) will reroute traffic to avoid preemption. Also, the incorporation of high technology, all digital equipment will reduce call establishment delays.

Increased security will be provided through use of the Secure Telephone Unit III (STU-III). The low cost STU-III will virtually replace every telephone in the military. DSN also provides service denial by appropriate guards against unauthorized tampering of switching equipment, transmission media and network management controls. Two areas that require special security are the unclassified software and DSN data base which are guarded against tampering by redundant methods. Furthermore, bulk encryption is used over all backbone lines, Administration, Operations, and Maintenance/Network Management (AO&M/NM) circuits, and Common Channel Signaling (CCS) circuits.

Cost effectiveness is achieved by using state-of-the-art digital commercial equipment and PTN services to the maximum extent possible. Since state-of-the-art technology will be used, long access lines will be shortened and DSN will allow the use of standard 12 push button telephones for precedence dialing without operator assistance. Also, almost all traffic currently carried over Wide Area Telephone Service (WATS), Direct Distance Dialing (DDD), and AUTOVON will be handled by DSN using most economical routing methods.

Interoperability will be extended by the development of interfaces with European Telephone System (ETS), Federal Telephone System (FTS), Japanese and Korean telephone systems and all public telephone systems. Both national and international standards will be possible. Consolidation of all DOD into one telephone system for secure and clear calls will further improve interoperability.

2. Features of DSN

DSN will provide all the features of the commercial telephone and several military-unique features. The CLASS features of the PTN will be available and include call forwarding, call waiting, and call transfer. Another feature is attendant camp-on which is used when preemption is not possible and the caller wants to get through to the destination as soon as possible but the destination end is busy. Therefore, the camp-on feature listens for the destination end to signal on-hook at which time it will automatically dial the number for the user, saving the user's time and frustration. DSN also will provide to designated users a malicious call identification feature. This feature is assigned by the Administration, Operations, and Maintenance (AO&M) facility on a line by line basis. [Ref. 42: p. 12-16]

Several features that will be available to high level users are nailed connections and conferencing. Nailed connections are simply semi-permanent connections based on the requirement of the user to maintain fast and preemption-free connection to the destination. The DSN will allow up to 10% of the switching capacity to be nailed. The other, generally speaking, high level feature will be the conferencing abilities. There are two basic types of conferencing that DSN provides: random and preset. Random conferencing can be further subdivided into two types which are progressive and meet-me conferencing. Progressive conferencing is where the user dials a precedence, a conferencing code, and then sequentially dials up to 19 other users through a conference bridge to form an ad hoc group. Meet-me conferencing is where the user will designate a specific time for conferees to dial an unlisted telephone number into a conference bridge to set up the conference. Preset conferencing is conducted by the user dialing a precedence and a seven digit number that routes the call to the appropriate switch and identifies the preset conference group. DSN will have the capability to maintain up to ten preset conference groups per switch. In addition, each switch has the capability to handle up to 30 meet-me, 30 progressive, and 40 preset conferences simultaneously. [Ref. 42: p. 15-39]

Since the DSN must support military requirements, some sort of precedence and preemption must be available to prevent critical communications from being blocked by lower precedence calls. Therefore, DSN will incorporate Multilevel Precedence and Preemption (MLPP) and Precedence Access Threshold (PAT). MLPP is nothing new in the sense the same basic concept is employed in AUTOVON today. The preemption is accomplished by the affected (point where blocking occurs) switching center sending a supervisory signal through the trunk, or in the case of a digital network, a message, through the common signaling channel toward both the calling and called user lines. These signals are received at the switches which release the trunk and send preemption tones of 440 and 620 Hz to the users. Today, AUTOVON prematurely preempts calls that will not be preempted with the DSN. These problems associated with MLPP will be solved by the increase in trunks, nodes, switches, better routing matrices, and overall efficiency of the DSN. PAT is another feature designed to overcome problems with preemption and will be discussed next. [Ref. 42: p. 28]

PAT is a software controlled mechanism that limits the number of simultaneous calls selected by user stations, trunk groups, or other groups. PAT sets an upper limit on the number of calls by precedence level and by the calling area. There are five precedence levels and five progressively wider calling areas that form a 25 cell matrix. A call is screened prior to actually entering the PAT and will be assigned a slot in the cell if limitations are not exceeded. If the limitation is exceeded, the call will be rejected. However, before rejection occurs, overflow to another cell may be allowed if additional conditions are met. The general rule for overflow is that if the sum of the threshold values of equal or higher precedence levels with wider calling areas is greater than the occupied slots of the cells of corresponding values; or if the sum of the threshold settings for all cells is greater than the sum of all calls in progress, the call will overflow and not be rejected. The idea behind overflow is to preclude unnecessary preemption. This feature was not available with AUTOVON. [Refs. 42: p. 27, 43: p. 5-10]

3. Phases of DSN

The general strategy of DSN is divided into two timeframes, near term and mid term. In the near term (FY90-91), the strategy focuses in on overseas acquisition and the implementation of DSN Phase I. The overseas nodes will be three to four times as numerous and the full network will be 10 to 20 times as large as the present network. Phase I of DSN will be complete when AUTOVON, AUTOSEVOCOM, and Defense Commercial Telecommunications Network (DCTN) are phased out in FY95. By the completion of Phase I, 59 nodal switches called multi-function nodal switches or stand alone nodal switches will be installed in the Western Hemisphere as seen in Figure 40 on page 96. [Ref. 41: p. I-4-1 - I-4-4]

In the Western Hemisphere, Phase I of DSN consists of the DCTN program, upgrades to Canadian AUTOVON, incorporation of AUTOVON into DSN, installation of three new nodal switches, and incorporation of the Defense Telephone Service Washington (DTSW) into DSN. In Europe, Phase I consists of upgrades to the European Telephone System (ETS), upgrade and incorporation of AUTOVON, and added CONUS-to-Europe connectivity. Europe will have a total of 45 nodal switches and 160 end office (EO) switches. In the Pacific, Phase I will include upgrades and incorporation of AUTOVON, Japan Telephone Upgrade (JTU), Korean Telephone Upgrade (KTU), Oahu Telephone System (OTS) upgrades and Scope Dial upgrades (Air Force telephone upgrade program). When complete, the European DSN nodes will be

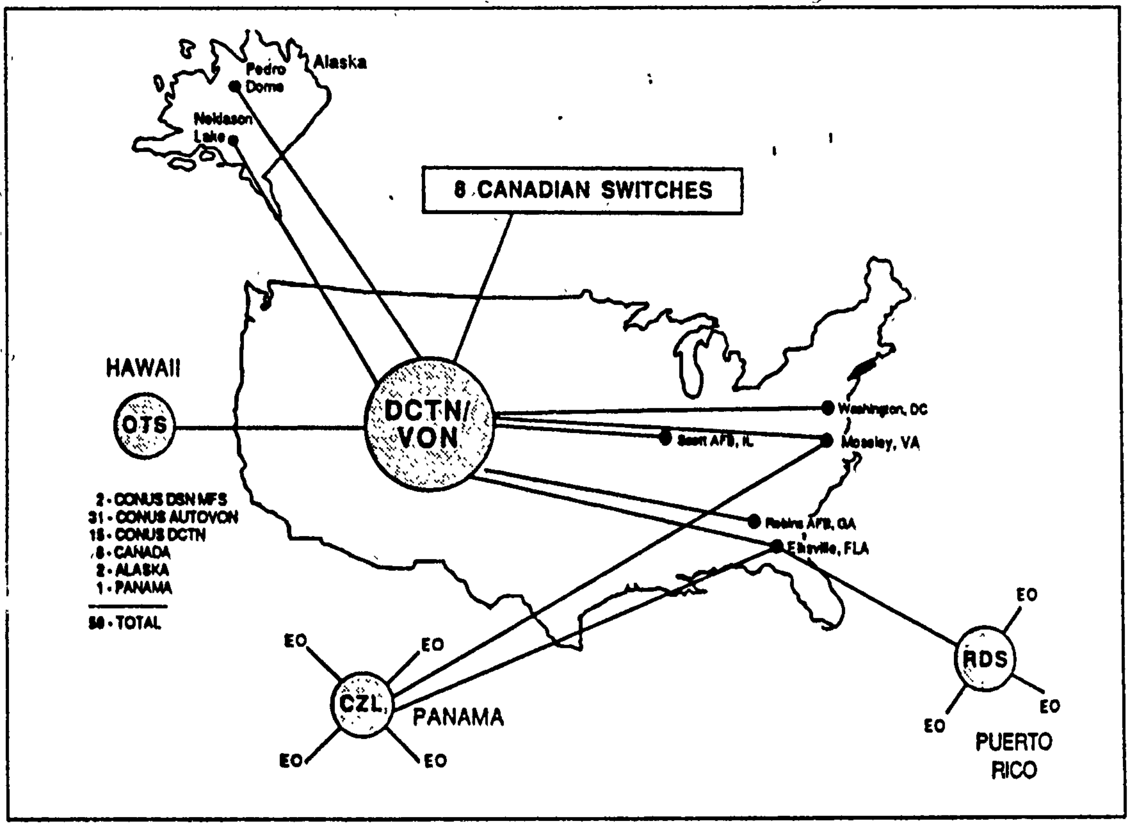


Figure 40. Nodal Switches in the Western Hemisphere [Ref. 44: p. V-1-2]

distributed as seen in Figure 41 on page 97. The Pacific nodes are depicted in Figure 42 on page 98 and will have 21 nodal switches and 48 EO switches. [Ref. 41: p. I-4-6 - I-4-7]

The mid-term (FY92-97) focuses on completion of DSN Phase II and initiation of Phase III. DSN Phase II will concentrate on integration of voice, data, video, and secure voice services into the network. The program that will implement these changes is called Integrated Defense Communications System (IDCS) and will consolidate DCTN, AUTOVON, and DSN phase I into one integrated system to provide better services and reduced cost. Phase II will be complete in FY97 when near real time control of the network is possible and key ISDN features, such as Common Channel Signaling (CCS), are employed throughout the network. Phase III (FY97 and beyond) will focus on a truly end-to-end digital network culminating in the capability of ISDN. [Ref. 41: p. I-4-1]

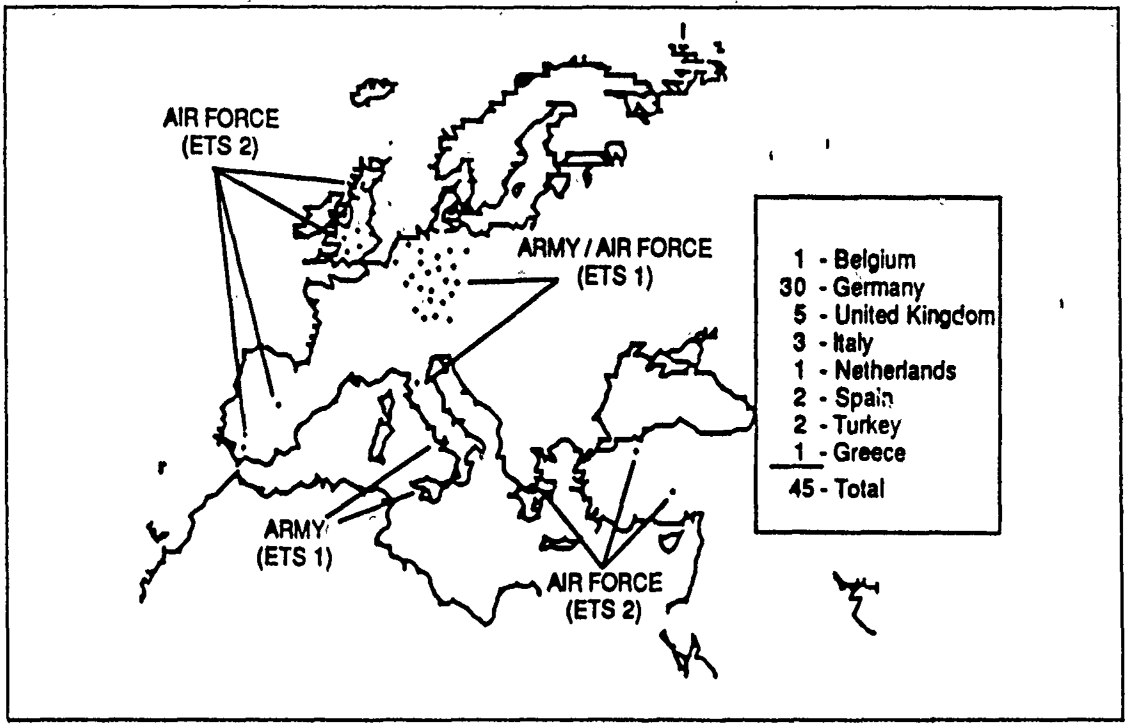


Figure 41. Nodal Switches in Europe [Ref. 45: p. III-1-2]

4. Architecture

The size of the network and reliability requirements produce an availability factor of .99996. As a rule, DSN is sized to meet 120% of the requirement today and will permit growth of 50% without the need for additional equipment. About the only two occurrences that DSN is not specifically designed to withstand are the effects of a nuclear blast and the associated electro-magnetic pulse (EMP), however, switches and equipment have been strategically placed to avoid the primary effects of these occurrences. Other sizing studies, such as the Flash Non-Blocking (FNB) study, helped to determine the architecture. The FNB study was conducted to determine the network sizing needed to prevent critical blocking of flash traffic in a crisis situation and also to ensure the physical diversity requirement for three separate media for inter-theater routing was achieved. Looking into the primary architecture reveals four major subsystems: [Refs. 42: p. 9 and 201, 41: p. I-2-6]

- Transmission and signaling
- Switching
- Timing and synchronization

- Administration, Operations and Maintenance/Network Management (AO&M/NM)

The transmission subsystem is being configured to widely disperse switching centers using both government owned and commercially leased transmission systems. The final transmission architecture will employ all digital facilities. A variety of transmission systems will be utilized such as microwave, satellite and fiber optics. Transmission requirements allow connection of the DSN to FTS, PTN, or other networks through an interface, but a connection through the second network to an additional network cannot be expected to provide an acceptable level of transmission performance. Since the switching in the DSN will be mostly commercially provided, care must be exercised in the signaling used in DSN. However, unnecessary constraints will only prevent the use of existing commercially available equipment. Additionally, signaling must be compatible with all host countries. Therefore, signaling in DSN will be conducted through the use of a Common Channel Signaling system number 7 (CCS7) modified for military use. One of the nice features of CCS is that premature preemption will be eliminated since CCS requires an end-to-end search of available trunks before seizure takes place. [Refs. 42: p. 145, 43: p. 5-1, 5-26, 6-11]

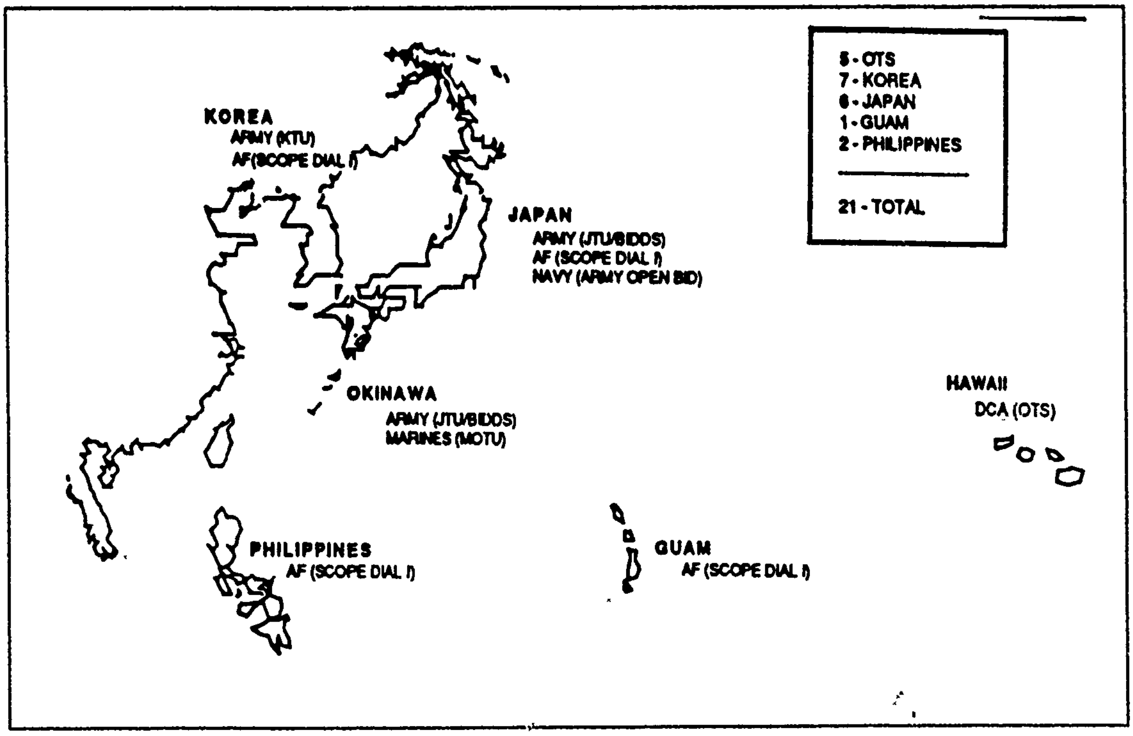


Figure 42. Nodal Switches in the Pacific [Ref. 46: p. IV-1-2]

The switching subsystem will consist of four types of switches: a multi-function (MF) nodal switch; stand alone nodal switch; end office switch; and a remote switching unit (RSU). The MF switch has the combined functions of a DSN backbone nodal switch and an EO and is comparable to a toll switch and a CO in the PTN. The SA switch has only the functions of the DSN nodal switch, or by comparison a toll office in the PTN. The EO switch will perform the functions equivalent to a CO in the PTN or a PBX for a large complex. The RSU provides service to areas where an EO is not economical and will feed to an EO or the EO port of a MF switch. Routing is determined by the switching facilities based on several criteria. First, satellite hops are restricted to two hops. Secondly, path selection uses a sequential routing algorithm that is essentially an exhaustive tree search selecting a path in a progressive manner. This means each link is chosen independently of the former link. If the node cannot find an available trunk, it will "crank back" the call to the preceding node and that node selects another path. Each switching system will have the capability to route calls over the primary route or up to nine alternate routes to the destination. If the node still cannot find an available trunk, it will preempt the lowest precedence call. [Refs. 42: p. 4-37, 43: p. 2-1, 4-1]

Another routing feature is the process of Most Economical Routing (MER) where the switch can automatically choose the most economical route based on a route cost level classmark. There are four levels of cost, and the switch will attempt connection at the lowest cost level first until all routes are exhausted. Then it will proceed to the next cost level and so on. Another feature of the switching system is that of dual homing. Dual homing is where the EO is homed on two separate DSN nodes which are accessible through a single telephone number. A call destined for that EO will attempt the first node. If busy, it will try the next node. If both are busy, then a busy signal is returned or preemption occurs.

The Timing and Synchronization subset is vital since the DSN requires synchronized clock rates based on a highly accurate frequency reference obtained from numerous sources including Global Positioning System (GPS), Long Range Navigation C (LORAN C), Cesium beam atomic clock or by connecting directly to other digital facilities such as AT&T, GTE, and MCI. For example, most of the DSN network in the U.S. uses timing from AT&T. Clocks at nodes are called station clocks and are slaved to LORAN C in the near term due to its wide range capability and ability to provide an accurate cesium beam at low cost with an availability rate of 99.7%. If LORAN C is not available, then the clock from the Defense Satellite Communications System (DSCS) is used. The clock from the GPS system will be used in the future. This means that the DSN nodal switches are plesiochronous or nearly synchronous in that there is some inherent slippage, but as long as updates can be received from the clock source, the difference is negligible. All MF, SA, and EO switches have internal clocks whose oscillators are synchronized by the selected source. RSU switches receive their inputs via DS-1 links by slaving to their EO switch. If for some reason the clock source is lost, the oscillators at each switch will be able to continue operation for up to 70 days. Each switch will contain two oscillators that use triple redundant synchronizing frequencies for reliability. [Ref. 43: p. 7-1, 7-12]

The final subsystem is the AO&M/NM subsystem which will be managed by DCA through a comprehensive computer support system that is survivable and secure. This computer system will automatically monitor and pinpoint system abnormalities, implement real-time switching to prevent network congestion and conduct continuous traffic analysis to optimize operations. The system has preplanned contingencies in the event any outages should occur. The computer managed system eliminates costly and delay producing human intervention.

AO&M/NM, as the name implies, can be broken down into the four subcategories of administration, operations, maintenance, and network management. Administration is the function that ensures configuration control over the system design aimed at specific requirements. Operations is the entity that deals with physical control and setup of the network. Maintenance is the part of AO&M that provides servicing and repairs to the network. Finally, network management conducts near real time analysis, collects network status reports and configures the network to optimize its response based on current network traffic conditions and equipment performance. The objectives of network management are to: inhibit switching congestion; utilize all available circuits by taking advantage of unused circuits due to time zone differences; keeping all circuits that have a high probability of success filled; and giving priority to calls that require fewer circuits to complete when the load is high. [Ref. 42: p. 159-165]

In support of these elements, there are six specific AO&M/NM functional requirements:

- Network management is the network supervision that allows near real time actions to occur in order to optimize the flow of traffic
- Maintenance aspects of the AO&M/NM subsystem service test the network and repair as necessary

- Resource implementation deals with the installation and acceptance of new equipment into the system and ensuring its compatibility
- Service provisions allow for installation of new services
- Performance surveillance provisions report on the actual performance of the network so that problem areas can be identified and fixed
- Administration, engineering and planning support the previous five requirements

The AO&M/NM subsystem is divided into a hierarchy for implementation on the Defense Communications Operations Support System (DCOSS). The DCOSS will provide the man-machine interface to control the network worldwide. In order to accomplish this, the network is broken down into theaters of operation composed of three basic elements. These elements are the Defense Communications Agency Operations Center (DCAOC), the Area Communications Operations Center (ACOC) and Alternate ACOC (AACOC), and the Subregional Control Facility (SRCF) which can be seen in Figure 43. Key locations in the hierarchy of the system will be located in Arlington, Va., Pearl Harbor, Hi., Vaihingen, FRG, Croughton, UK, Clark AB, PI, and Yokota AB, Ja.. The entire Western Hemisphere will be contracted with military administration elsewhere. [Ref. 41: p. I-B-4]

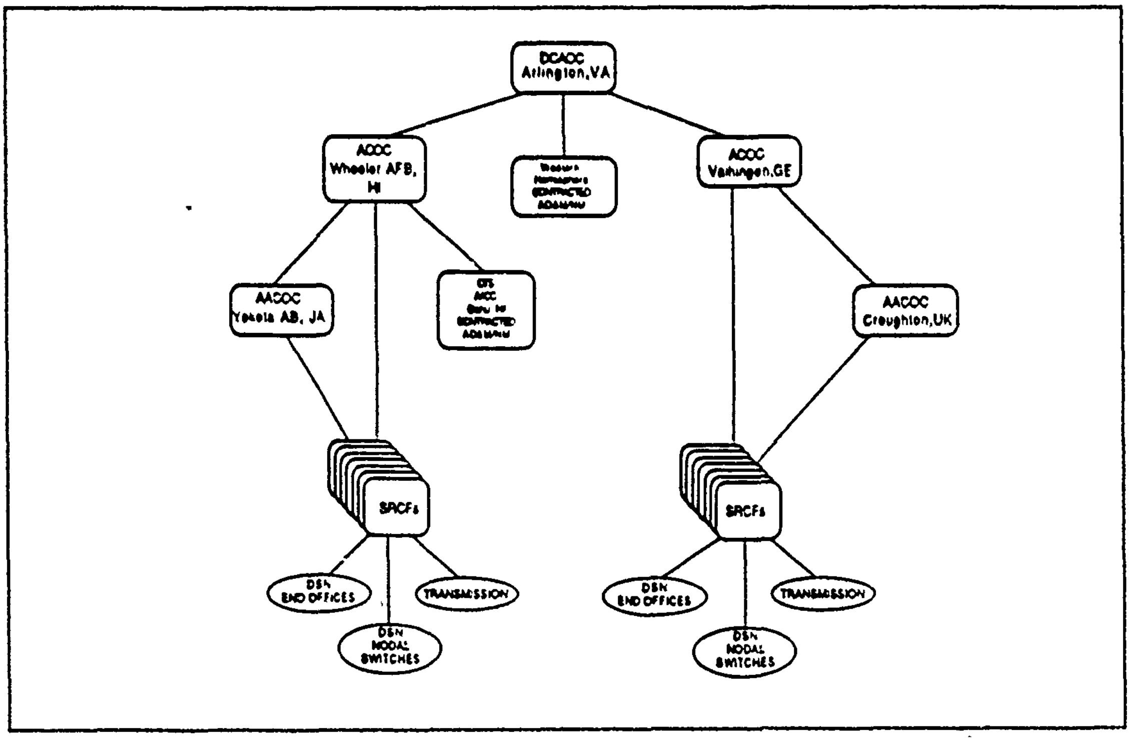


Figure 43. DSN AO&M/NM Hierarchy [Ref. 41: p. I-B-4]

5. Usage Based Billing

The DCA will plan, program and budget for acquisition and leasing of the DSN based on user inputs. These user inputs will be used to compute a subscriber rate so each military department can pay its share of the backbone network. Subscriber rates are based on services that can be provided, even though they may not have been used. The rate structure is centered around Maximum Calling Areas (MCA) that are defined below: [Ref. 41: p. I-5-3 - I-5-5]

- Global service which is unrestricted to more than two theaters
- Inter-theater service which is restricted to two theaters
- Theater service
- Area service
- Local service which is usually restricted to a post or base

The usage based billing scheme is designed to make the cost causer pay. If the users do not want to spend as much on telephone service, then they must submit a requirement for fewer services. These requests for services can be submitted to DCA quarterly, however, DCA can only adjust the size of the network annually since it takes one year to redesign the network. The design work is mostly done by the Defense Communications Engineering Center (DCEC). In either case, DCEC requires an annual requirements validation to avoid a "wish list" situation. About 80% of all DSN costs are related to trunking. Therefore, overestimations by the user causes the sizing of the trunks to be excessive. [Ref. 41: p. I-5-5 - I-5-8]

The usage based billing process begins when the Defense Commercial Communications Office (DECCO) receives inputs from the military departments specifying requirements in terms of MCAs, precedence, and amount of traffic in erlangs. DECCO formulates the bill and passes it to the military departments so that they can develop their billing scheme for their end users. The actual bill is then sent to the military departments every month based on actual usage and the planned bill. Each military department must now analyze its bill to ensure it is not wasting assets. If the services provided are not adequate or more than needed, the military department can notify DCA to adjust its requirements, but it must realize there will be a one year lag in sizing the network. [Ref. 41: p. I-5-5 - I-5-8]

One last note on usage based billing is that it is not the same as usage sensitive billing which has been in the evaluation phase since 1987. If usage sensitive billing becomes a reality, billing will reflect per unit of time charges sensitive to distance and time of day. This is what the commercial sector uses today. [Ref. 41: p. I-5-3]

6. Worldwide Numbering and Dialing Plan

The numbering plan for DSN is similar to commercial telephones today, except they contain extra digits for military requirements. The basic parts of the DSN Worldwide Numbering and Dialing Plan (WNDP) are listed below.

- Access digit is basically a preparatory command
- Precedence digit assigns one of five precedence levels
- Route digit informs the switch of special routing or termination instructions used when connecting to another network
- Route control digit is used to control special features
- Area code equates roughly to a theater code
- Switch code equates to a specific EO
- Line number equates to a specific terminal(s)

The decision to use abbreviated dialing in overseas theaters rests with the Unified CINC. The use of additional features is the local base commander's decision. Four digit dialing may be used if the projected switch growth is less than 7000 lines and five digit dialing can accommodate up to 10,000 lines. Any requirement over 10,000 requires use of the full number as more than two switches will be required. [Ref. 43: p. 3-1, 3-5]

7. Makeup of the Defense Switched Network

The DSN is made up of many different parts. It will use the latest technology and be redundant in coverage. Using the Pacific theater as an example, there are numerous submarine cables crisscrossing the ocean between CONUS, Hawaii, Guam, Japan, Korea, Wake, Midway, the Philippines, and Okinawa. In addition, Figure 44 on page 104 and Figure 45 on page 105 show the satellite connectivity between points in the Pacific theater. The overall big picture for telephonic communications media is shown in Figure 46 on page 106 which shows the redundancy of the overall system. Similar redundancy is used in the other theaters.

Since DSN is a very expansive program, it will affect many other programs directly. Several of these programs will disappear as DSN becomes fully operational with respect to that program's capabilities. Programs that are essential to DSN are listed below. [Ref. 41: p. I-A-1, I-A-12]

- European Telephone System (ETS) which will replace the three existing systems in Europe

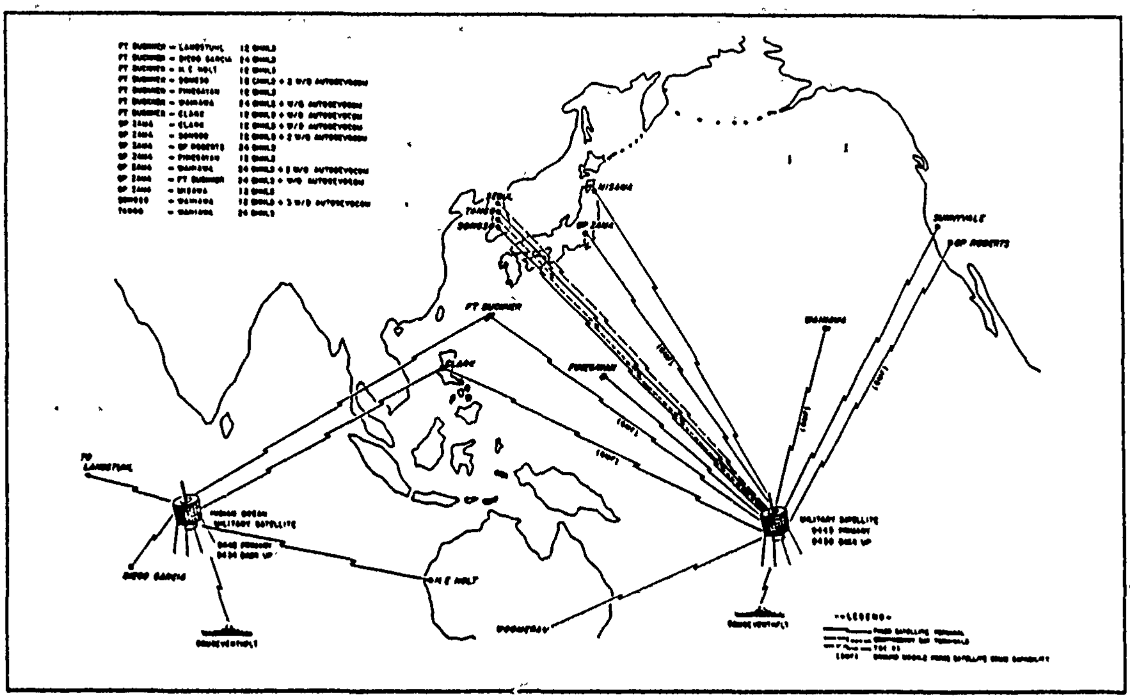


Figure 44. MILSAT Connectivity in the Northwest Pacific Region [Ref. 47: p. 46]

- Korean Telephone Upgrade (KTU)
- Scope Dial which is a program to purchase digital switches for overseas elements of DSN
- Base Information Digital Distribution System (BIDDS) which is a restructuring of Scope Dial and several other Air Force programs into one
- AUTOVON which has been in service since 1961 and will be phased out in 1995.
- Oahu Telephone Service (OTS) which will provide all military telephone services on Hawaii
- Japan Telephone Upgrade (JTU)
- Nationwide Emergency Telecommunications System (NETS) which will provide essential government communications after a national emergency using surviving resources of the Public Switched Network (PSN) and DSN and will be operational in the early 1990s
- Defense Commercial Telecommunications Network (DCTN) which is a 100% leased program and forms a major portion of DSN phase I in CONUS
- Defense Satellite Communications System (DSCS) which will provide a reliable, anti-jam protected transoceanic segment of DSN
- Numerous transmission programs for overseas upgrades to digital transmission
- Commercial carriers which will form the backbone of DSN both overseas and CONUS

- Secure Voice System (SVS) which will replace AUTOSEVOCOM in the early 1990s. It consists of the Secure Telephone Unit III (STU-III) program and the Red Switch Project (RSP)

Additionally, DSN must interface with many other programs that include:

- NATO Initial Voice Switched Network (IVSN) which is the NATO equivalent to AUTOVON
- Federal Telecommunications System (FTS) which is presently using a 1963 vintage system that is obsolete and will be replaced by the FTS-2000 program that will be for all government users. The Assistant Secretary of Defense (ASD C3I) made the decision for the Department of Defense (DOD) that the DOD would rely on DSN for its needs. However, under the terms of a DOD-GSA memorandum of understanding, the DOD will evaluate FTS-2000 when it becomes operational and contracted in 1990 and will compete FTS-2000 services against other service providers.
- Defense Metropolitan Administrative Telephone System (DMATS) which is a management effort to consolidate all unsecure telephone services in a region under one manager
- Integrated Services Digital Network (ISDN) which will form the basis for DSN end-to-end digital connectivity
- Foreign Post, Telephone and Telegraph Exchanges (PTTs) which are the interfaces to various host nation commercial networks

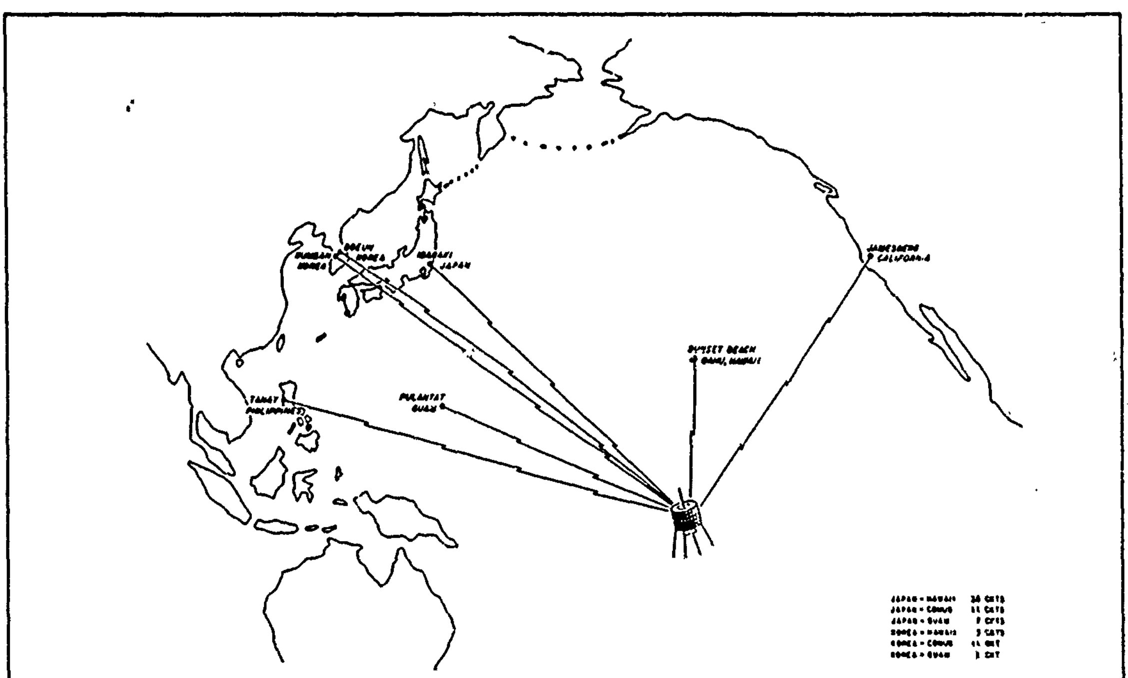


Figure 45. Commercial Connectivity in the Northwest Pacific Region [Ref. 47: p. 53]

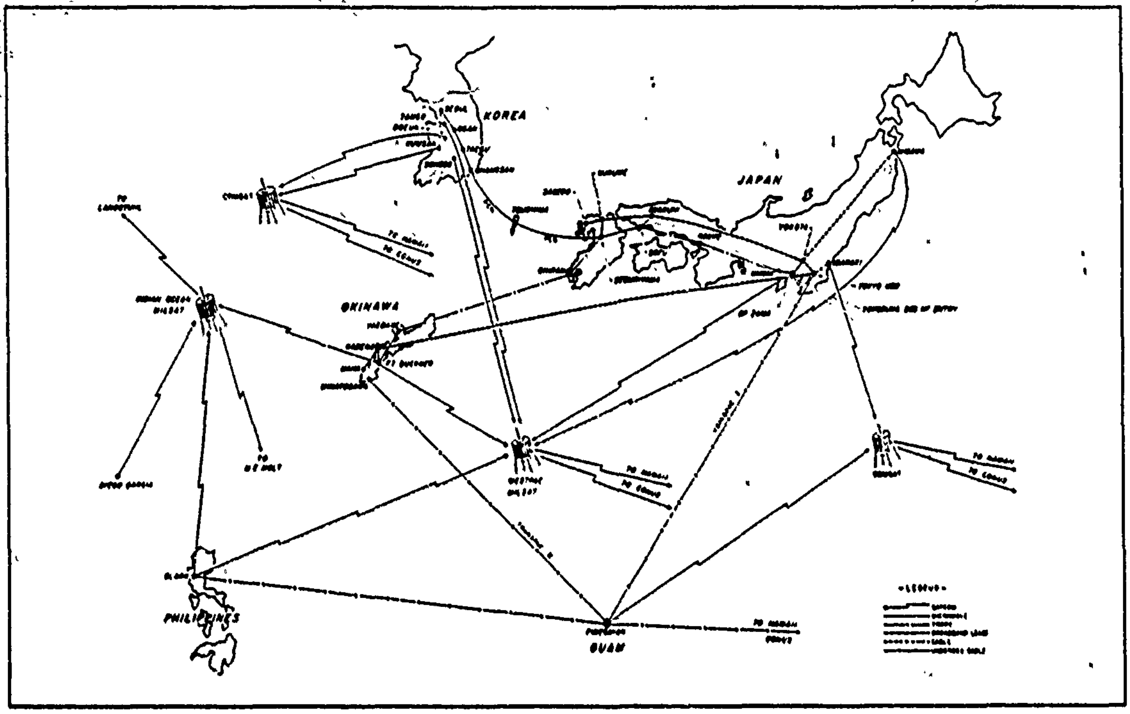


Figure 46. Communications Interconnectivity in the Northwest Pacific Region [Ref. 47: p. 8]

- AUTODIN which is the military's message switching network
- Defense Data Network (DDN) which is a survivable packet switching data network

8. Defense Commercial Telecommunications Network

A major division of the DSN is the Defense Commercial Telecommunications Network (DCTN). The DCTN represents part of the DSN segment for CONUS, Alaska, Hawaii, and Puerto Rico. The key role of DCTN is to offload routine traffic from the AUTOVON network in the short term. By the end of FY90, an estimated 42% of the routine AUTOVON traffic will be supported by DCTN. It will incorporate all of the DSN features and will be more robust due to the availability of many commercial telephone systems in the U.S. it can utilize. The system that DCTN will replace uses 1961 vintage, obsolete, analog equipment that is difficult to repair and actually costs twice as much to make a regular telephone call as using the new DCTN system. [Ref. 44: p. V-3-4]

The DCTN is quite complex using 28 service nodes and over 272 general user locations. The key elements of the service nodes consist of a satellite earth station, a specially designed 5ESS switch, encryption equipment and the Digital Access and Cross

Connect System (DACS). The relationship between DCTN and AUTOVON is seen in Figure 47 on page 107. Note that this figure actually represents the DSN phase I architecture for the Western Hemisphere. The system will be controlled by an AT&T staffed complex located in Dranesville, Va. Inter-nodal links will travel via terrestrial or satellite links with maximum of one hop. The entire network is many times redundant. [Ref. 48: p. 58-65]

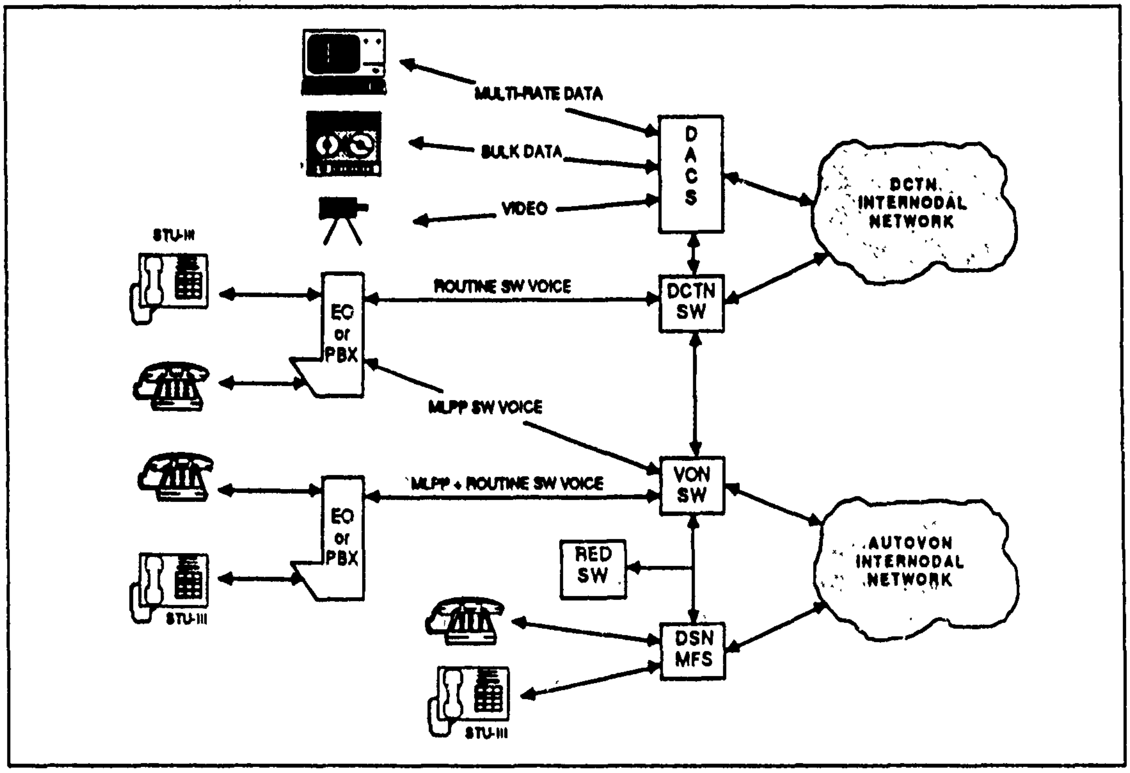


Figure 47. Relationship Between DCTN and AUTOVON [Ref. 44: p. V-1-3]

C. FEDERAL TELEPHONE SYSTEM

The Federal Telephone System (FTS) will be a key element in DOD telephone communications beginning in 1990 when cutover to the new FTS-2000 system begins. DOD will cutover to FTS those services it feels are more economically provided. FTS will be able to provide to the military non-precedence services including direct distance dialing, 800 service, WATS, and other services not provided by the defense communications system. FTS is being managed by the General Services Administration (GSA) on a multi-billion dollar contract. The contract calls for leasing the system for ten years and price reviews every three years. The initial up front service initiation charges the Navy will have to pay are only two million dollars. [Ref. 49]

The FTS program is split between two vendors, AT&T and U.S. SPRINT, providing 60% and 40% of the network, respectively. FTS will support 1300 user locations nationwide, including Puerto Rico and Guam. The network is actually divided into two networks, the "A" network is the AT&T network and the "B" network is the U.S. SPRINT network. The "A" network will have 18 #5ESS toll switches and will be a mixture of fiber, coaxial, and satellite media. The "B" network will be all fiber optic cable, but will have fewer switches. [Ref. 49]

The architecture of FTS will provide what the industry calls a P07 grade of service. This means that seven percent of all calls will be blocked on the initial try. In comparison, DSN ranges from P00 for flash override to P10 or higher for routine calls. The backbone lines for FTS will be provided by AT&T and U.S. SPRINT and will connect to the FTS switches. The FTS switches will be directly connected to the military EO, PBX, or satellite earth station by new trunks laid by the vendor. The number of trunks between the switch and the EO determines the grade of service. One exception to this is for bases that use less than 720,000 CCS per month. In this case, there is no direct connection. Instead connection to the military facility will be provided by the LEC company. This type of service is called virtual on net service. [Ref. 49]

D. SECURE TELEPHONY

1. Background

The two networks that presently provide secure telephony are the AUTOVON network and the AUTOSEVOCOM network. The AUTOVON network is quite familiar to us and provides secure voice communications when encrypted at each end. The AUTOSEVOCOM network is actually two subsystems composed of a narrowband and a wideband subsystem. The narrowband's primary limitation is its poor voice quality. The wideband's primary limitation is its cost. Every wideband channel requires the equivalent bandwidth of 12 normal telephone calls. In addition, both of these subsystems are old and have cumbersome switching systems that support them and have considerable duplication in their networks without gaining reliability or survivability, just increased cost. Wideband presently employs the KY-3 encryption device and the KYX-9 telephone unit. The entire network has only 1800 users worldwide. The advantage of the narrowband subsystem is that it requires only one telephone channel. The narrowband subsystem uses KY-57/58 Vinson or KY-65 Parkhill encryption equipment. However, the voice quality, as mentioned earlier, is poor, somewhat akin to listening to someone talking underwater who sounds like "Donald Duck." As a final comment on the AUTOSEVOCOM system, the name is actually a misnomer in that the system is far from automatic. Many switches are manual switches called Secure Cord (SECORD) switchboards and are installed at many locations worldwide. Even some high level commands are supported by manual SECORDS, such as USCINCEUR. [Ref. 50: p. 41-66]

Another system in use today is the Secure Telephone Unit II (STU-II) which uses narrowband technology that provides better voice quality than the narrowband portion of AUTOSEVOCOM. However, the STU-II has some definite limitations since it weighs 70 pounds and is considered communications security (COMSEC) equipment and must be stored in a vault or secure area. Additionally, the cost of the STU-II ranges from $16,000-$18,000 per unit. However, the unpopularity of the STU-II was primarily driven by differing service needs. The Navy preferred upgrades to the existing AUTOSEVOCOM system over the STU-II so that it could retain interoperability with the fleet while the Air Force and Army preferred the STU-II. [Ref. 50: p. 41-60]

In 1983, the National Security Agency started to take a hard look at COMSEC within DOD. It was especially concerned with Telephone Security (TELSEC), and when it finished, its report was summed up in one word - "dismal." The existing AUTOSEVOCOM system was developed and built in the 1950s and was obsolete, relied on expensive leased circuits and served a very limited number of subscribers. Although the STU-II program had begun in 1975 and was just starting to deliver the first telephones in 1983, NSA concluded that both the AUTOSEVOCOM and the STU-II combined were not even remotely capable of solving the problem. Like AUTOSEVOCOM, the STU-II was expensive and served a limited audience of approximately 10,000 users. The NSA analysis showed that by the mid 1980s, the requirement would be closer to a half million users! There was an additional problem of keeping the civilian sector, especially DOD contractors, from discussing classified data over the telephone. To NSA, the situation was obvious: a whole new initiative in TELSEC was needed. The answer was to develop a telephone unit that would meet every user's needs, be affordable to all levels, be reliable as a normal telephone, and be available in the near term. In just over two years after NSA made the decision to proceed with the project, industry developed the STU-III with R&D seed money from DOD and a substantial amount of its own money. [Ref. 51: p. 18-22]

2. Secure Voice System

The Secure Voice System (SVS) is the secure telephonic system of the future. It is comprised of the Secure Voice Improvement Program (SVIP), Secure Conferencing

Project (SCP), and the Red Switch Project (RSP). The SVS will not be operational until 1995, and secure voice will remain on the AUTOSEVOCOM and AUTOVON networks until they are phased out. The SVS will be carried on the DSN with an interface to the Jam Resistant Secure Communications (JRSC) system for critical high level communications. The JRSC uses the DSCS instead of the DSN. The SVIP program consists of the STU-III Low Cost Terminal (LCT), a cellular STU-III, and the Key Management System for the STU-III. The STU-III, which is the backbone of the SVS, is being manufactured in basically two models, a type I and a type II. The type I is designed for classified information and the type II is designed for unclassified, but sensitive information. The price for the units ranges from $2,000 to $5,000. The physical size of the STU-III is similar to that of a commercial telephone recording machine and is ready to hook up right out of the box. The crypto package is self contained and the unit does not need special installation, requiring only a standard telephone jack and a single 110 Volt outlet for power. The STU-III encrypts and decrypts in the unit itself and requires no special transmission media, therefore enabling the use of standard telephone lines so its signal can be sent over all telephone lines worldwide. [Ref. 50: p. 41-60]

The STU-III is used just like a regular telephone. If you want to place a secure call, simply dial the number as you would a regular telephone. Once the party you want answers, either end can press the secure button, the telephone automatically authenticates in less than 12 seconds, and you're ready to talk.

The nice feature of the secure mode is that the voice quality sounds normal without all the static and "underwater comms" of some of the past systems. In addition, all STUs have another security feature called a Crypto Ignition Key or CIK. When the CIK is removed, the unit is unclassified, requires no special storage, and functions as a normal telephone. When in the secure mode, a display panel is activated on the phone indicating what level of security is authorized based on security clearances of both parties. The display also indicates who the caller is by name or by position. Both of these features are provided by the code on the CIK which can only be used in the STU for which it was programmed. [Ref. 51: p.18-22]

All STUs are capable of handling Top Secret telephone calls, including Sensitive Compartmented Information (SCI). The only restriction is the level of access to which the person assigned to the CIK is cleared. Finally, as with all secure communications, there has to be some crypto or keylist. Again, the simplicity of the STU-III is overwhelming. Once the initial unit is "loaded," it requires only an annual update which is done by calling one of several toll-free numbers to a key management center. The entire process takes about five minutes using an encryption process called "firefly" which changes with every call. Additionally, the unit itself is no longer considered COMSEC and does not need special handling. The unit is controlled by a serial number, and only the CIK is considered to be sensitive. If the unit is tampered with, the crypto inside will automatically zeroize. [Ref. 50: p. 57]

There are some limitations to the STU-III which center around NATO interoperability and shipboard communications. To cure the NATO problem, a STU-IIIA has been developed to be compatible with the NATO STU-II. In order to retain shipboard connectivity, the Navy has retained the KY-3 program until the Advanced Narrowband Digital Voice Terminal (ANDVT) is operational and will replace the KY-3 units.

The second element of the SVS is the RSP which is simply a PBX located within a secure compound. The purpose of the red switch is to provide command and control users secure voice within their command centers through a network of 12 switches located in the National Military Command Center (NMCC), Alternate NMCC and the 10 Unified and Specific commands. A Red Interface Terminal (RIT) will provide the interface between the DSN and the red switch. Red switch trunks will use bulk encryption. [Ref. 41: p. I-2-16]

The final segment of the SVS is the SCP which is a jam resistant, secure conferencing system for about 40 high level users. SCP uses the DSCS to provide these services and is directly connected to the red switches. It will have the ability to conference up to 20 users including afloat users.

3. Base Information Transfer System

Currently, Navy bases both overseas and in CONUS lack the facilities to smoothly handle telecommunications needs. The existing Navy base communications capability consist of separate networks having very little if any interoperability as shown in Figure 48 on page 113. By taking just a quick look at this figure, it is easily seen that communications systems have evolved into separate systems along the way. The BITS system is designed to eliminate this problem by consolidating voice and data with long haul and local information systems between shore-based information systems. This can be seen in Figure 49 on page 113 which depicts the target BITS architecture. Note that at the heart of BITS is the fourth generation, all digital PBX that will provide the necessary switching of voice and data for both intra-base and inter-base communications. The other part of the BITS program is to upgrade base cabling to handle all voice and data requirements. This will normally be done through the use of fiber optic cable. With this architecture, the end user will be totally interoperable with all the communication systems and will allow ships to plug into the system when pierside. BITS will be installed in three phases. Phases I and II will be completed by 1996 and will provide most of the services. Phase III represents the target period in which BITS will transition to the ISDN architecture after 1996. [Ref. 52: p. 1-2, 5-8]

The target architecture for BITS is ISDN. However, BITS is part of a larger architectural plan called the Navy Data Communications Control Architecture (NDCCA). NDCCA has basically three elements which are BITS, long haul communications systems, and afloat communications systems (includes internal shipboard comms). The NDCCA target architecture is shown in Figure 50 on page 114 which depicts two BITS systems being connected by the long haul networks. The long haul networks currently in use are DSN (AUTOVON), DDN, and AUTODIN. Target long haul networks will be DSN, DDN, and Digital Patch and Access System (DPAS) and will comprise the Integrated Data Services (IDS) system. Note that BITS will provide ships in port pierside plug-in capabilities or access through the Naval Telecommunications Center (NTCC). [Ref. 52: p. 2-8]

4. National Security Emergency Preparedness Programs

National Security Emergency Preparedness (NSEP) programs fall under the National Communication System (NCS). The NCS is an interagency group assigned to provide NSEP telecommunications throughout the full spectrum of national emergencies. There are three NSEP programs that deal directly with the telephone system and will be discussed in the next sections. They are the Nationwide Emergency Telecommunications System (NETS), Commercial Network Survivability (CNS), and Commercial SATCOM Interconnectivity (CSI) programs. Together they form the National Level NSEP Telecommunications Program (NLP). [Ref. 53]

a. Nationwide Emergency Telecommunications System

The National Emergency Telecommunications System (NETS) is a system that is designed to provide essential interoperable communications for federal agencies during and after a national emergency. The types of communications it will provide are voice and low speed data. NETS first came to mind during the 1970s when studies concluded that the PSN would, for the most part, survive an attack. Therefore, AT&T was called upon to investigate the possibilities of an embedded structure within the PSN. The study concluded that NETS was technically feasible and that some transmission augmentation was necessary. As a result, the NETS concept was approved in 1988.

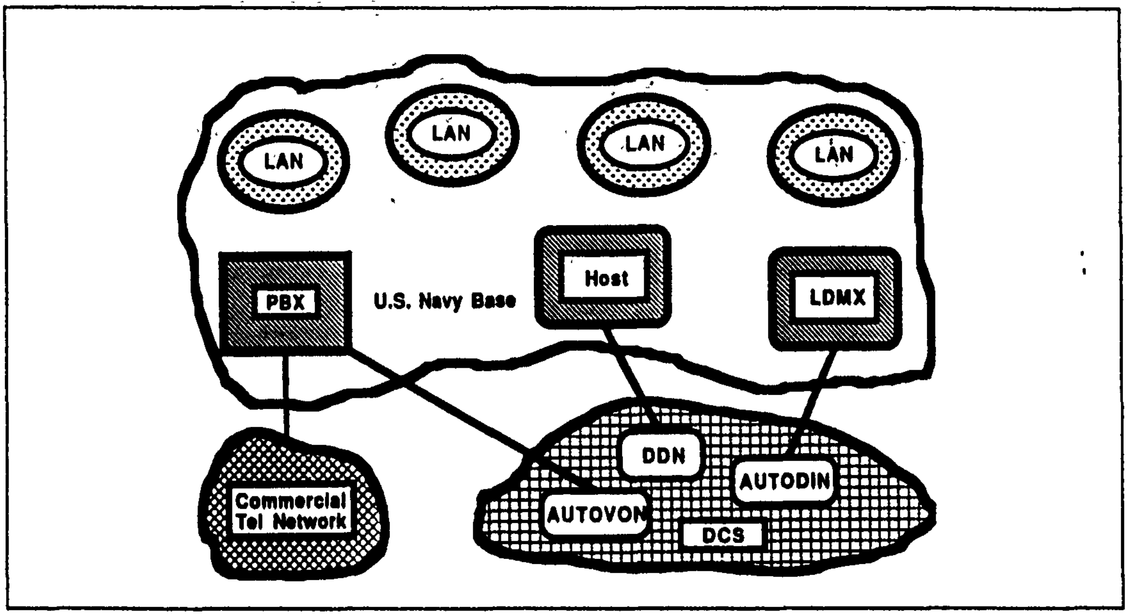


Figure 48. BITS Current Situation [Ref. 52: p. 4-2]

The basic strategy of NETS is to take advantage of all the PSN resources, including AT&T, MCI, Sprint, BOCs, and Independent TELCOs as well as the resources of the DSN and FTS-2000. The network is composed of three elements. First is the Call Controller (CC) which is just a switch that has appropriate NETS features added

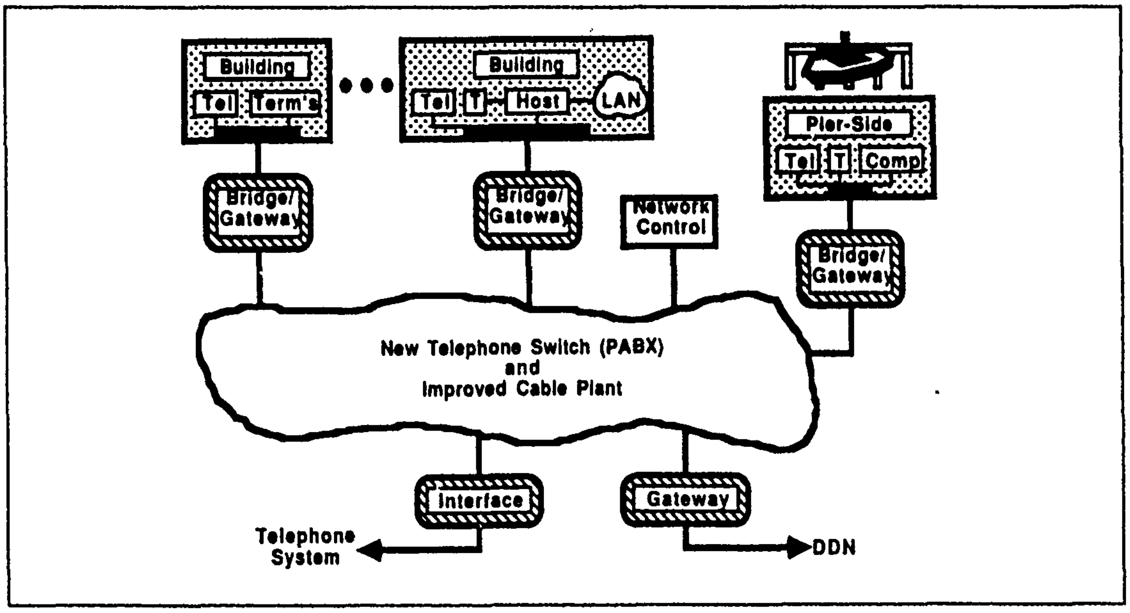


Figure 49. BITS Target Architecture [Ref. 52: p. 4-3]

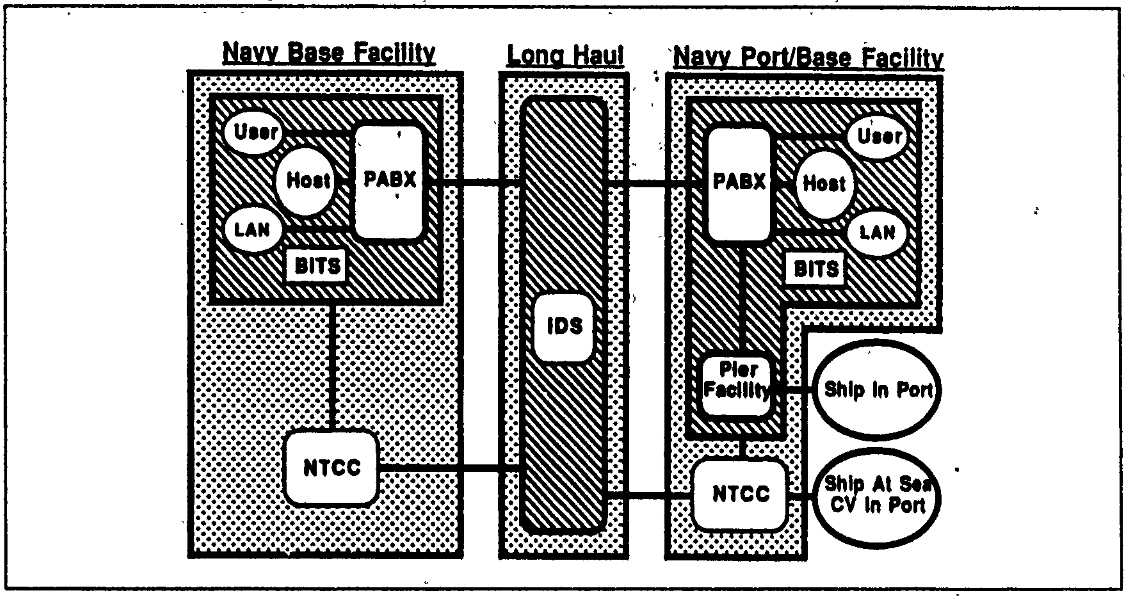


Figure 50. **Navy Data Communications Control Architecture (NDCCA)** [Ref. 52: p. 2-9]

to it. Next is the Access Security Device (ASD) which is a handheld device that allows the user to identify himself and set a precedence. The ASD comes with a keypad and display screen to enter information and connects directly to the earpiece of the telephone to initiate the call. Finally, there is the NETS Maintenance and Administration Center (NMAC) which supports NETS operations.

To initiate a call over NETS, a user picks up the ASD and enters the proper precedence. A NETS user has automatic precedence over normal PSN traffic and has the ability to preempt lower precedence calls for some users. Once the number is dialed, the CC switches search for a route and pass control to each switch as it progresses. Unlike the PSN, NETS will have more routing options available and will progress through a much more rigorous search before a "crank back" occurs. Because of the unlikely chance of a blocked call, NETS will provide the reliable and survivable network needed for crisis management. [Ref. 54]

b. Commercial Network Survivability

The Commercial Network Survivability (CNS) program consists of five elements: data base acquisition and maintenance; NSEP survivability; network management; carrier interconnect; and Mobile Transportable Telecommunications (MTT). Data base acquisition allows DOD the ability to simulate various battle scenarios and assess the network. NSEP survivability aspects will provide certain commercial telephone installations with backup power supplies, physical security, hardening, and additional route diversity. Network management will provide the means to differentiate NSEP calls, reduce dial tone delay, alleviate congestion, and provide additional dual homing. The carrier interconnect aspect will connect commercial carriers not presently connected at various locations to provide a more robust interconnection of the networks. Finally, the MTT component will provide mobile interconnection for the network. The concept for MTTs will allow disjoint pieces of the network to be "spliced" together if congestion or a disconnect should occur. [Ref. 55]

c. *Commercial SATCOM Interconnectivity*

Commercial SATCOM Interconnectivity (CSI) is the last of the NSEP programs. The purpose of CSI is to interconnect AT&T 4ESS switches to ensure commercial satellite connectivity in a crisis. Studies showed that most earth stations for commercial satellites were lacking diverse routing from the earth station to the network interconnect. CSI will provide 20 key earth stations with the requisite diverse routing. In addition, CSI will provide emergency Telemetry Tracking and Command (TT&C) facilities in the event commercial TT&C facilities fail. [Ref. 56]

E. CONCLUSION

The commercial telephone system is the evolutionary result of over one hundred years of historical and regulatory events described in the first two chapters of this thesis. Some of these events were a result of technological advances and some can be attributed to simple politics of the era. In either case, the telephone has come a long way since the beginning and will continue to be the primary means of communication for the commercial sector as well as the military. Finally, the military is very dependent on the commercial sector for its telephony requirements and must continually assess commercial advances in technology and regulatory hindrances to ensure the best telephone network is being provided.

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