

NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

Public Switching Telephone Network

Graduation Project
EE- 400

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Lefkoşa - 2002





TABLE OF CONTENTS

ACKNOWLEDGEMENT'	i
ABSTRACT	ii
INTRODUCTION	iii
1."INTRODUCTION TO TELECOMMUNICATION & PSTN	1
1.1 History of Telecommunications.	1
1.1.1 Broadcasting. Human Voice.;	1
1.1.2 Television	2
1.1.3 Computer and New Media	3
1.2 Networks	4
1.3 Public Switching. Telephone Network (PSTN)	6
1.4 Telecommunication Concept	8
1.4.1 Analog and Digital Networks	9
1.5 Switching	9
1.5.1 Service Evolution	
1.5.2 Technology Evolution	11
1.5.3 Network Switching	13
1.6 Advanced Switching	15
1.6.1 Space and Time Division	15
1.6.2 Time Division	15
1.6.3 Frequency Division	16
1.6.4 Technologies	17
1.6.5 Space Division Switching	17
1.7 Control Methods	20
1.7.1 Hierarchical Control	20
1.7.2 Regressive Control.	21
1.8 Step-by-Step; & Switching System	22
1.9 Band Switching System	23
2. TELEPHONE SET	24
2.1 Telephone Set	24
2.1.1 Basic Functions of Telephone Set	24

2.1.Z	The Basic Elements Of Ille I.elephone.Set T.clephone	25
2.2	T~le_pho:ne l'io,letw-0:rk	11
2.2.1	How Telephone Works	27
22.2	Why 48V voltaige is m'eed in telep-hane systems--?	29'
223	What is .sea.Jing Current?	29
2.2-A	Wh:y-FuU fiuip-lex{~er-a!ivtn in Single Wire Pair?	:10
2.25	What'is:.fue< bamtwidth- o,f the' teleph-011e' liner	J0
2.J	Network Interface Details	
2.4	S'i;gnaling	3.2
2.4~1	Mmm.JIW:-:"Xclrame Sigl\almg	'!2
2.42	Signal Eurietinns.	33
2.4 .3	Suhser.fue r;,.f.,@el-f'l"S-rgurail-ing:	33
2.4A	later Office Signaling	34
24.5	C'5f}tnmon-Chmme1 :f!fltemffice Signaling	35
2.4.6	Telephone.Number;ing Plan.	37
2.4.1	Loeal-Loop- Signaling De.sign	37
2.5	'lelephone Line Parameters	JS
2:S.1	Line Balance	38
1.S.2	Loop. carreas.effecis,	3&
2.h	Netwnrk fut:erfa:ee·mTe-fephcwm	19
2..6.1	Simplified Traditiena] Network foterface	39
2.6.2	wh:Y oaioon mict-0phone in telephones ?	41
2.6.1	Typical, El;;;,r0Iµean,,Nea,v~11k,	41
2Ji,4	Simplified ILR. Siaadard "425,R"	41
2 li.5	Details ofw:iri:ng inside telephones used in USA	43
2.7	Telecem Hybrid Ci'fcuits·ForOther Equipment Then Telephone	44
2 '1.~1.	T~z-z~r,1;·t~"" "<> ~~~~~·~o11o~~~~ hub i-l-~~~~~P.M.P.M.M.M.P.	44-
2.8	Siemens andITT Resistive Hyh:rids	45
2.8.J	One T.ransfo.rmer Hybr&d Circuit for 2 Wires to 4 \.V-ires Conversion	-46
2jt2	·:Onnnectin:g ~Hyh:rid:unincl ilelephone ·Equipment"	47
2.8.3	Simple Intercon.r:r:1:ectioa.:with.no P:owet'ProVrd'.etlttrtheLine'	47

2~8.4 Simplest Powered Ciceit	47
2.8.5 :Gemmd Hyhrid Interface	48
2.8.6, C{}mponmts for Telephone. Li;ie.lmerfaeing	49
2_9 Telephone, Ljae DetaHs i-ii, Di-fferen,t_ Cotint:l:fes.	SQ
2.9.1USA	S.1
2.9.2 Fin:land	51
2.10 Notes: ameut lel:epho:ne 'frtnafO'fimeis1	52
3.. €6MPARISONS OF COMMUNICATION' MODALITIES AND"	
MEDIA	53
3.1 V-oice and Video	
3.2. Text amt Image	54
3..3- Text an1Speech-	55.
3.4 Total Traffic	55
3.5 Phfups Roomtling Internct	57
3.6 Norcim Re.cording: Interlace	5&
.1.7 Recording Interface From Tekni:ikm1-Maailma Magazine.	59
3.8 Marantz PMD Recorder Series	60
3_9 Telephone to Studio Mixer Interface	
3.H1 Audi©- Jntetfaes- without l'ira:nsf©;umer. lseilatinn	
3_11 Simple: Tcleeo1n Hybrid: Citenit.1{.	62
3.12 Modified Circuit	64
3.13 Souridcar1 T-0 Teleph{')ne { .ine Interface	66
3..14 Opration1'al, Ampiliier Based Hybrid Cire,aits	67
J_15 More Details Implementing Telephonr I .ine Interface:	70
3.15.1 "\Vet" transformer	70
3.15.2 "Dry" transformers	
3.16 Problems in Linking. Teleph@otte Mybrid·t@- Aud~ System	13
3.16..1 I,eve:!" Matching on Lrical a.ml: Remote Voiee	73
3.162 Irans~hybrid loss .aad .anaonacer voice distortion	73
:U6.3 Echo Prublem .in Long Distance Catls	74
J.J6-_4 MetaHic Sour:ding Caller-Voice Pmblem	75

1.17 Helpful Tips for Telephone Hybrid -Circuit Designers	75
1.17.1 Measuring the Return Loss of a Cable	75
3.1.1 Simulating telephone line	77
1.1.1 Resistor and capacitor network simulation models	17
1.1.2 Building a Telephone Line Test system from a Telephone Cable	78
3.19 Testing Stuntwires	79
3.19.1 Other Technical Regulations for Telephone Test Equipment	79
3.19.2 Equipment in Series with Telephone	80
4.1 SOURCES OF FAILURE IN THE PSTN	
4.1 Failure Classification	83
4.2 Finding	84
4.3 Observations	86
4.4 Why So Reliable?	87
4.4.1 Reliable Software	81
4.4.2 Designing for Reliability	88
4.5 Loose Coupling	89
4.6 Human Intervention	90
CONCLUSION	92
REFERENCES :	94

ACKNOWLEDGEMENTS

First of all I would like to thank my supervisor Prof Dr. Paklreddin. Mainedov upon his attitude towards my project. He guided me as much as he can and with all his efforts I was able to complete my Graduation Project. He provides me a great knowledge about this project and also about the communication field. His words of encouragement kept me going, and under his supervision I made this graduation project and it is a writing good experience for me. I took 2 subjects about communication and got good grades because when I was in the lecture I felt myself learning under the world of knowledge. His way of teaching is so kind that I could understand whole lecture at the moment I chose this project because he has good **knowledge on communication.**

I am very grateful to my friend Malik Osama Nazar, who helped me in this project. He is nice and gentleman and always he helped me in my any difficulties. Also I am thankful to Malik M. Sajio and, Naveed Akhtar because they spent time for me and helped me to search some Information from Internet. These both friends have a great impact on my heart. I wish a peaceful life for them and a happy life,

At the end I am paying regards to my parents, who help me on every stage of my life where ever I need, and they pick me up about studies and in all matters of life, and it is only because of their prayers that I am completing my degree.

ABSTRACT

Public Switching Telephone Network (PSTN) has a great importance and is used in every important official & public work. PSTN is also used in Army communications, it also plays an important role in field of science. With PSTN, it became possible to connect the whole world through Internet, which keeps it in touch with every part of the world. A telecommunication system, can take many different forms. PSTN has a very important role in our life:

The PSTN is one of the most important aspects of information theory, upon which many of technological advancements in Communication.

Public Switching Telephone Network is now part of our environment. Everyday we receive and transmit information by this network, often without knowing it.

Because of these and more, I prepare to choose my project's subject as "Public Switching Telephone Network".

PSTN is very wide field, and it cannot be covered even by one book. So we can find lot of books talking about this subject and each book has its own point of view;

One of the main objectives of this project is to give the reader enough of understanding to ask him or her the right question.

As I am doing this project to cover the important subject such as a wide field. Whatever, the assessment one can be assured that PSTN will continue to occupy an important place as a mean of communication.

INTRODUCTION

Telecommunications is a term for a broad field that can include transmission through satellite, microwave, telephone, and computer networks. Public Switching Telephone Network has a great importance and used in every important field like in official & public work.

PSTN is also used in many communications. It also plays an important role in fields of science. With PSTN, it became possible to connect the all over world through Internet, which keeps in touch with current affairs of the World. A telecommunication system can take many different forms, PSTN has a very important role in our life.

In Chapter 1 we briefly describe the major elements and types of communication device, switching systems in telephones and telephone network. Previous and Modern history of Telecommunications and importance of PSTN in communication and in Life span.

In Chapter 2 there is a discussion about Telephone Set that how telephone works and how the systems layout.

In Chapter 3 we briefly described which type of interference would involve in telephone system. How the long and short distance calls can be transferred and which main factor is involved.

In Chapter 4 we briefly described the failure of PSTN and which factors would be involved in the Public Switching Telephone Network (PSTN).

To aid in the volume of traffic between toll centers and primary centers, sectional centers and regional centers are used. The best route is shortest route or the route utilizing smallest number of switching centers.

1. Introduction to Telecommunications & PSTN

Telecommunication is a term for a broad field that can include transmission through satellite, microwave, telephone, and computer networks. In order to transmit information from one computer to another, one must have access to a telecommunication system that maintains a large computer server with connection facilities. Universities, commercial telecommunications system providers and community networks have connection facilities. There are three primary methods used for establishing connection:

- 1) Dial-up connections with modem and a regular telephone line (Terminal Emulation). From this kind of connection you need a telecommunications software package (Procom, Zmodem, Versa term, etc...) installed on your hard disk.
- 2) PPP (Point to Point Protocol) connection with high speed modem and a regular telephone line. This kind of connection provides access to all multimedia features of internet. To establish this kind of connection, you need to install special PPP software, and TCP/IP (Transmission Control Protocol & Internet Protocol) software on your hard disk.
- 3) Direct connection through local area network.



1.1 History of Telecommunications

1.1.1 Broadcasting the Human Voice; ..

Since telephone wires could carry the human voice, could the wireless broadcast speech sounds as well? In 1906, a Canadian scientist named Reginald Fessenden converted sound waves into a pattern, or signal, in the radio waves. This pattern altered the amplitude, or height of radio waves. This method of changing radio waves was called amplitude modulation. We use AM for short:



Figure 1.1

By the early 1920s, regular broadcasts were being made in many "Countries. Radio was a great success. People everywhere could keep in touch with the latest news and listen to children's programs, concerts, or plays. In 1939, the American engineer Edwin Armstrong discovered a way to improve the signal further. It is known as frequency modulation, or FM.

1.1.2 Television

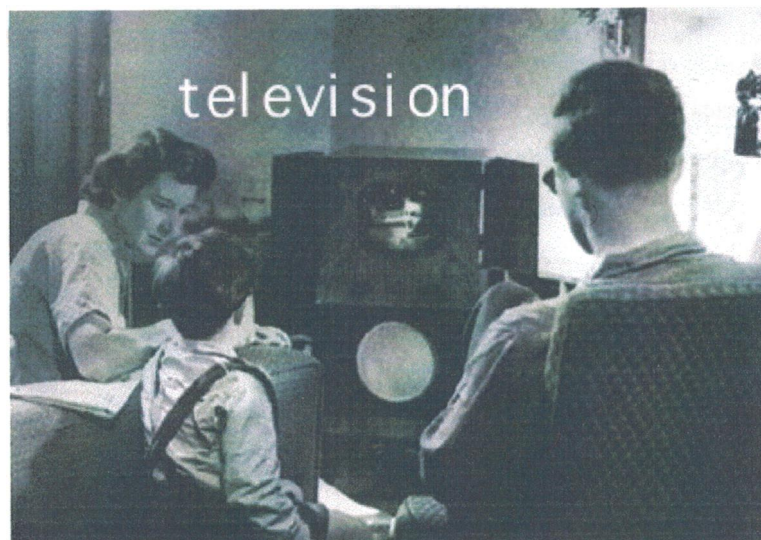
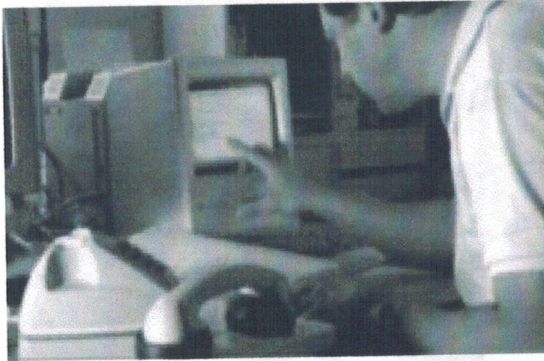


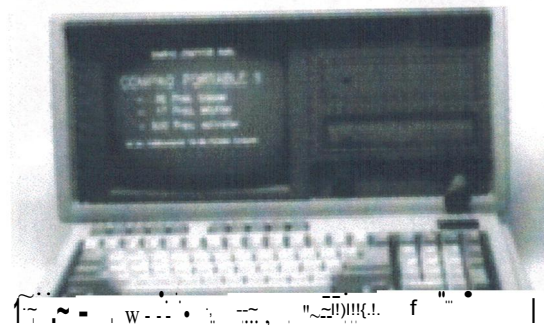
Figure 1.2

In December 1883, a young German named Paul Nipkow had thought of a way to send a moving picture by wire, He knew that a substance called selenium lets more electricity pass through it in bright light than when it is in the dark. He thought he could use this fact to turn a picture into an electrical signal,

1.1.3 Computer and New Media



Modems and acoustic-docking-stations. More and more people are using modems and acoustic-docking-stations. Speed: 300-



First real Laptop. Compaq launches the first portable AT (with batteries)



PCMCIA-cards. The PCMCIA-standard is going to be the most common standard for mobile communication and online hardware.

Despite being very complex, global telecommunications service is comprised of

1.2 Networks

Figure 1.3

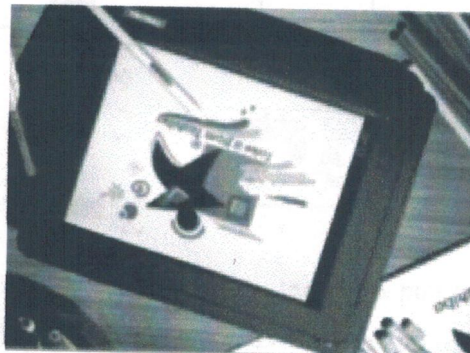
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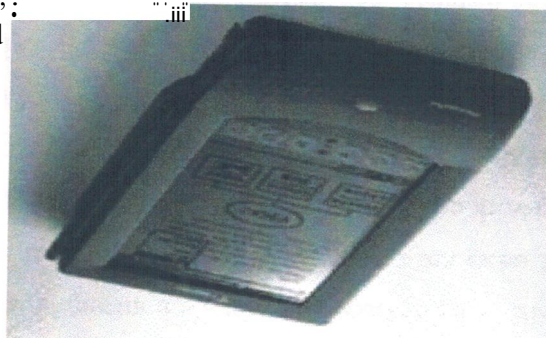
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a few basic network components, which are: (1) user equipment telephones, computers, and all the other devices that provide a means of accessing the network; (2) the access network users are connected to the main network by wire line or radio links; (3) the main network copper wire, microwave radio, and optical fiber cables connecting all the nodes of the global network; (4) transmission equipment the means by which huge volumes of information (there are many millions of telephone and data calls made every second) are carried over the network; and (5) switching equipment the hierarchy of local, long-distance, and international switches that allow any user of the network to connect to any other user. Each of these components has to consist of a combination of hardware and software.

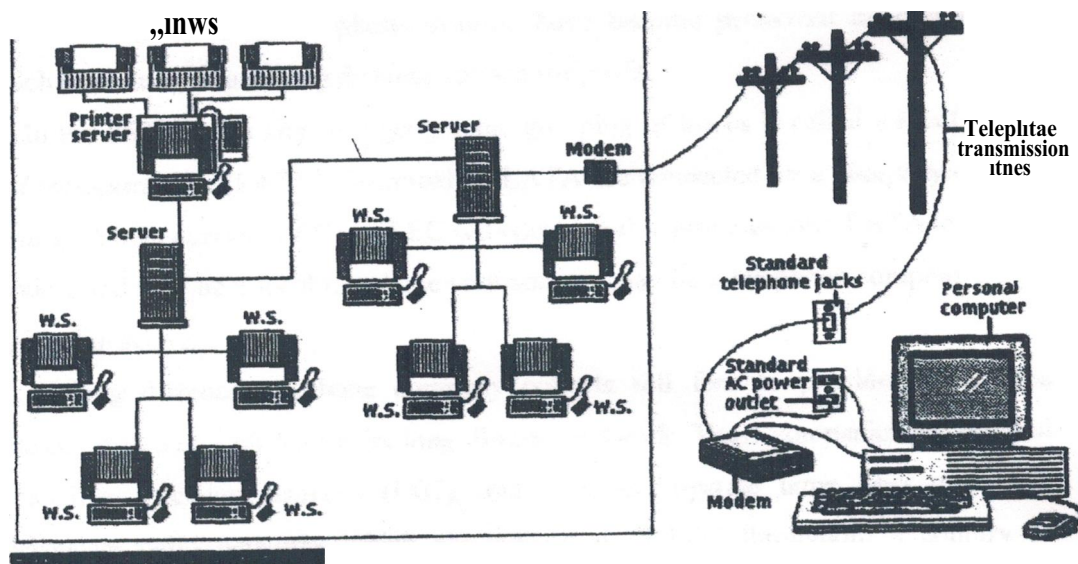


Figure 4

Hardware

This usually covers items such as telephones, transmitters, cables, interface devices, switches, and computers. In the past, telecommunications have relied heavily on hardware, such as dedicated switching elements, and on the logic providing its control functions. A situation is now developing in which more of the system relies on elements operating under computer (software) control. Because this software can be upgraded, this makes it easy to add new, enhanced functionality later.

Software:

This is code that instructs a computer or network device. Until the 1980s, most of the operational instructions used by a telecommunications network were hard-wired or

·pPe\$et. The advent of digital systems and data networks has led to a much wider range of network services, Software solutions are well suited to the complexity and flexibility inherent in these services,

1.3 Public Switching Telephone Network (PSTN)

The PSTN is a highly integrated communications network that connects over 70% of the world's inhabitants. In early 1994, the International Telecommunications Union estimated that there were ~650 million telephones in the world, as compared to 10 million in 1970. While landline telephones are being added at a 3% rate, wireless subscriptions are growing at a greater than 50% rate. Every telephone in the world is given calling access over the PSTN.

Each country is responsible for the regulation of the PSTN within its borders. Over time, some governments have nationalized by nationalization & which provide local and long distance service for profit.

In the PSTN, each city or a geographical grouping of towns is called a *local access and trunking office* (LAT). Surrounding LATs are connected by a common call-ed: a *focal exchange* (hEC); A LEC is a company that provides inter LAT A service, and may be a local telephone company, or may be a telephone company that is regional in scope,

A local distance telephone company collects toll fees to provide connections between different LATs over its long distance network. These connections are referred to as inter exchange carriers (IXCs), and own and operate fiber optic and microwave radio networks which are connected to LEC throughout a country or worldwide.

Figure (3) is a simplified illustration of a local telephone network, showing a local exchange consisting of a central office (CO) which provides

Figure (L5) is a simplified illustration of a local telephone network, called a local exchange. Each local exchange consists of a central office (CO) which provides PSTN connections to the customer premises equipment (CPE) which may be a residential phone at a residence or a private branch exchange (PBX) at a place of business. The CO may handle as many as a million telephone connections; The CO is connected to a tandem switch which in turn connects to the VSTN. The tandem switch connects the local telephone network to the point of presence (POP) of the long distance carrier or more IXC. The POP is the point where the long distance carrier's network meets the local network.

connect directly to the CO switch to avoid local transport charges levied by the LEC.

figure (LS) also shows how a 'PBX may be used to provide telephone connections throughout a building or campus. A PBX allows an organization or entity to provide intermd calling and other rrr-binding services (which do not involve the LEC), as well as private networking between other organizational sites (through leased lines from LEC and IXC providers), in addition to conventional local and long distance senders which pass through the CO. Telephone connections within a PBX are maintained by the private owner, whereas connection of the PBX to the CO is provided and maintained by the LEC

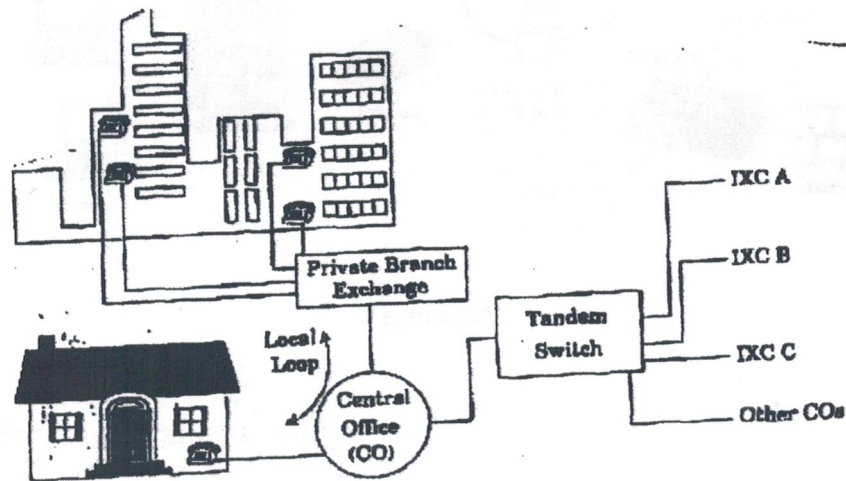


Figure 1.5

Since the invention of the telephone the public switch telephone network (PSTN) has grown proportionately with the increase demand to communicate. Switching services beyond metropolitan areas were soon developed increasing the size and complexity of the central office. New methods of switching were required to interconnect central offices through the use of interoffice trunks and tandem trunks as shown in figure. When the call is made outside the local area, they are routed through Toll Trunk and Toll Center.

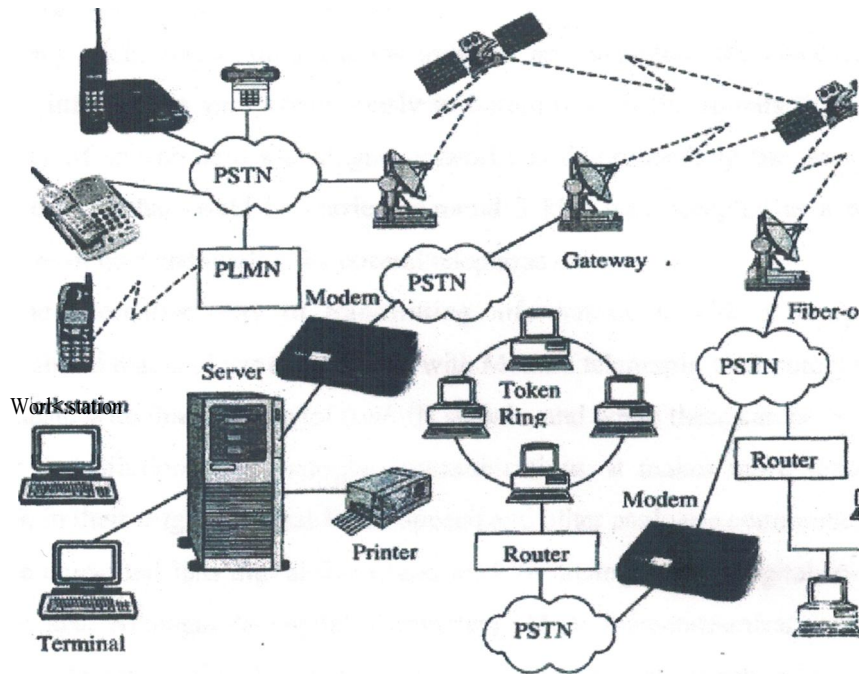


Figure 1.6

1.4 Telecommunication Concept:

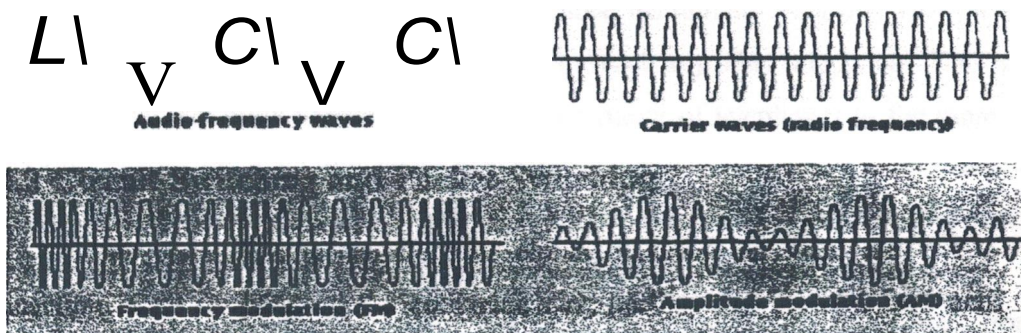


Figure 1.7

There are several ways of carrying information between senders and users. The options chosen should reflect the type of communication required. For instance, humans compensate for noise and transmission errors when they talk to each other. Unexpected delays or echoes cause problems in understanding, however. Computers have the reverse characteristics being tolerant of short delays and less so of transmission errors. The following concepts underpin telecommunications networks.

1.4.1 Analogue and Digital Networks:

Many of the telecommunication systems are analogue; the electrical signals conveying information vary continuously in harmony with the sounds they represent. The quality of speech across analogue networks is determined by the amount of the speech spectrum that could be carried. Around 3 kHz was accepted as a reasonable compromise of cost and quality for normal telephone calls.

The alternative way of transmitting information is with a straightforward electrical signal, that is either on or off, as with Morse's telegraph. Computers also communicate with discrete, digital (on/off) signals, and while these can be converted to tones for transmission over analogue communications, it makes more sense to send them back in their original digital form. Speech and other analogue communications can readily be converted into digital form, and back again (see: Digital-to-Analogue Converter and Analogue-to-Digital Converter). Most telecommunication networks today are "integrated" digital systems, ideally suited to computer networking and other multimedia applications, such as speech (voice), data, text, fax and video.

1.5 Switching.

1.5.1 Service Evolution

As originally invented by Bell, the telephone communication went from a particular telephone, instrument to only one other telephone instrument. This was the "private-line" service (Figure 1.5(a)), and there was no way to reach any other telephones. This private-line service was soon extended to connect a number of telephones to the same line, a "party line". Everyone could hear everyone else hence there was no privacy, and one call would prevent anyone else from using the line. Clearly, there was the need for a telephone to be switched to any other desired telephone.

One way to perform this switching would be to bring lines from all telephones to all other telephones. A switch at each telephone would then make the connection with the appropriate line to the desired telephone (Figure 1.5(b)). If the universe of telephone stations, to be reached is small, such a system could be used. However, if the universe is large, then the large number of lines that must terminate at each telephone makes such a system of station switching impractical.

The next solution was discovered only a few years after the invention of the telephone by Bell. The solution was a centralized switching system. The first of these was the "central office" (CO) which was a building containing a large number of switches. Lines from all telephones in a local area would terminate at the CO. Lines from the CO would then be switched to the desired telephone.

to a common place where the electrical connections were made to connect one station to another. The actual connections were made by people. The central place where the lines all came together was called the "central office" or the "exchange."

As exchanges grew to cover greater geographic areas, it became uneconomical to bring lines from the more outlying areas to one central office. More central offices were created, each serving a nearby surrounding area. Connections between central offices were made on lines called trunks (Figure 1-8(e)). As growth continued, special switching offices were developed to handle the trunks between a numbers of central offices. This centralized switching of trunks was performed at a switching office called a tandem office. New switching needs developed to serve long-distance or toll circuits between cities. Switching offices were thus devised to perform the switching of toll trunks only. These offices were called toll offices (Figure 1-8(g)). For businesses, a fair amount of telephone traffic was between telephones that were all located on the customers' premises. These telephones, therefore, could be served most efficiently with a private switch located on the premises. This switch was called a private branch exchange, or a PBX—a term that is still in use today. Present PBXs are automatic (PABX), using electromechanical or electronic technology (EPABX).

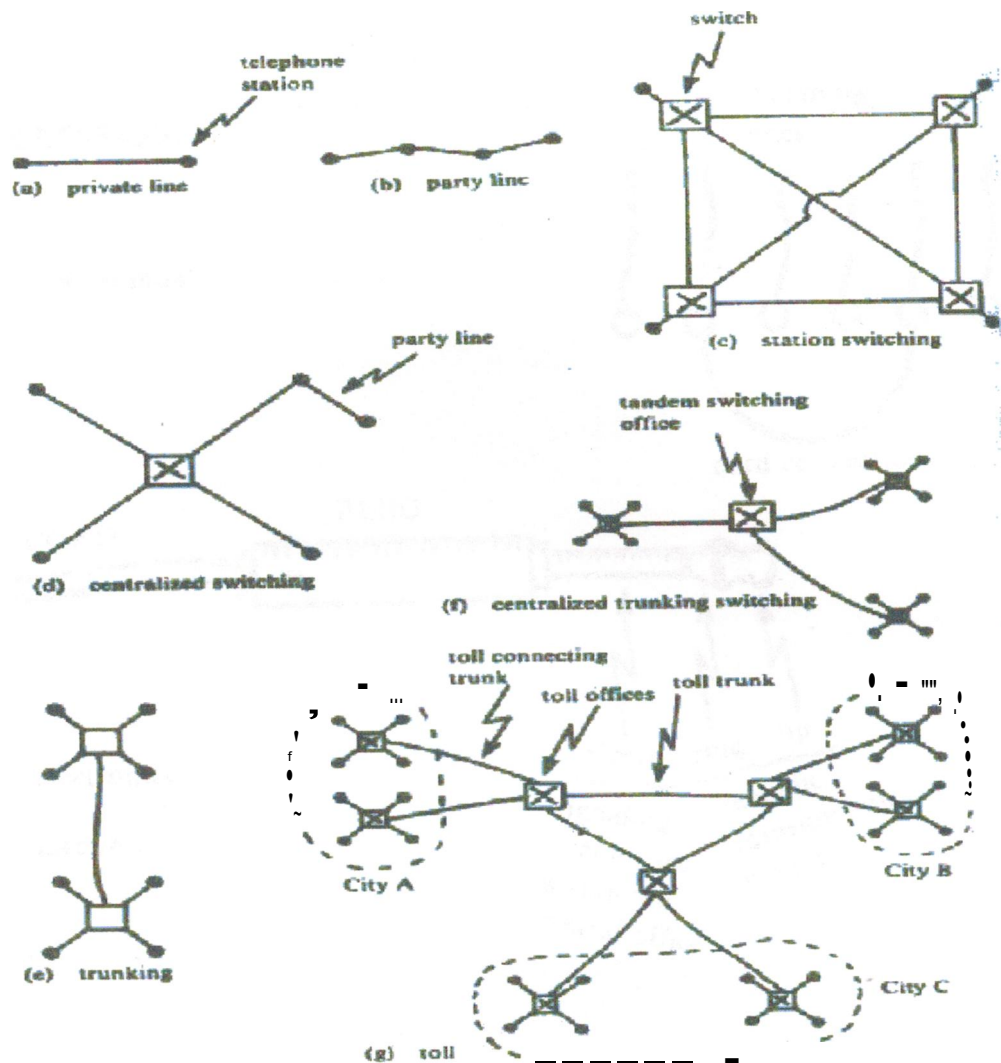


Figure 1.8

1.5.2 Technology Evolution

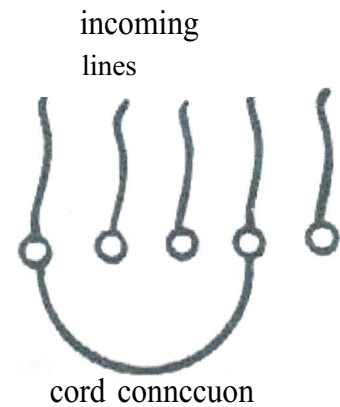
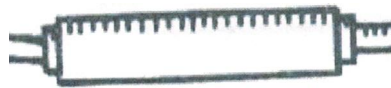
During the first few decades of telephone communication, switching was a manual operation performed by human beings who made the actual connections of circuits. (Refer to Figure 1-9.) The connections were made at a switchboard utilizing cords with plugs at the ends. The plug had a tip and a ring, which made the actual connection between the lines. The sleeve was used for signaling, and supervisory purposes, in common-battery exchanges. The terms "tip" and "ring" continue to be used to this day for the two wires between the central office and the actual telephone instrument. The circuits desiring service and the availability of trunks were indicated by small lamps.

PROGRESSION:

- Manual --

CORD

PLUG



- Electromechanical

- Electronic

- analog

- digital

signaling
circuitry used
within the
Central office

Figure 1.9

The first major innovation in switching came in 1892 with the first installation of an automatic switch, controlled by the telephone instrument itself. This switch was conceived by Almon B. Strowger. A later modification of this system included the invention of the dial and the use of dial pulses to control the operation of the switching system. The Strowger switch was an electromechanical device. Strowger's invention was adopted for use by the Bell System in 1919. Bell System engineers later developed improved automatic switching systems using electromechanical technology. The electromechanical technology was somewhat slow, not very flexible in terms of offering new services, and frequently generated electrical noise in the connection.

The current generation of technology for telephone switching is electronic using either microprocessor or digital switching techniques. The electronic technology is extremely fast and flexible. There are two major parts of any telecommunication

switching system: the switches themselves and a control section, as shown by Figure 1-10. The switches form the switching network. The control section operates the switches at the proper time in order to make a communication connection.

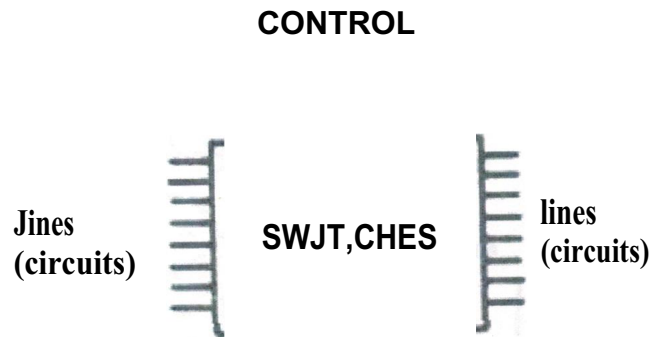


Figure 1.10

1.5.3 Network Switching

The switching network is composed of a number of centralized locations called switching offices, where the switching or connecting of circuits is actually performed. These offices are organized into a hierarchy dependent upon whether a call is local (exchange) or long-distance (toll). The three major types of switching offices are local, tandem, and toll. The local office is the one closest to the telephone station and connects directly to the local loop. A tandem office serves a cluster of local offices. Toll offices, or toll-centers as they are sometimes called, are concerned with long-distance toll connections.

There are five classes of offices in the Bell System switching hierarchy. At the lowest level is the local office or end office. This is the switching office where the call first originates and finally terminates. The end office is classified as a class-S office. A class-S office will attempt to make the connection directly to the terminating class-S office over a direct interoffice trunk. If necessary, the switching capabilities of a tandem office may be used. If the call is a toll call, then the toll switching offices become involved.

The *class-5* offices connect to class-4 offices, or toll centers. Toll centers connect to class-3 offices, or primary centers. Primary centers connect to class-2 offices, or sectional centers. Sectional centers connect to class-1 offices, or regional centers. The whole system is organized in a tree fashion with regional centers at the top of the tree, as illustrated in figure 1.1

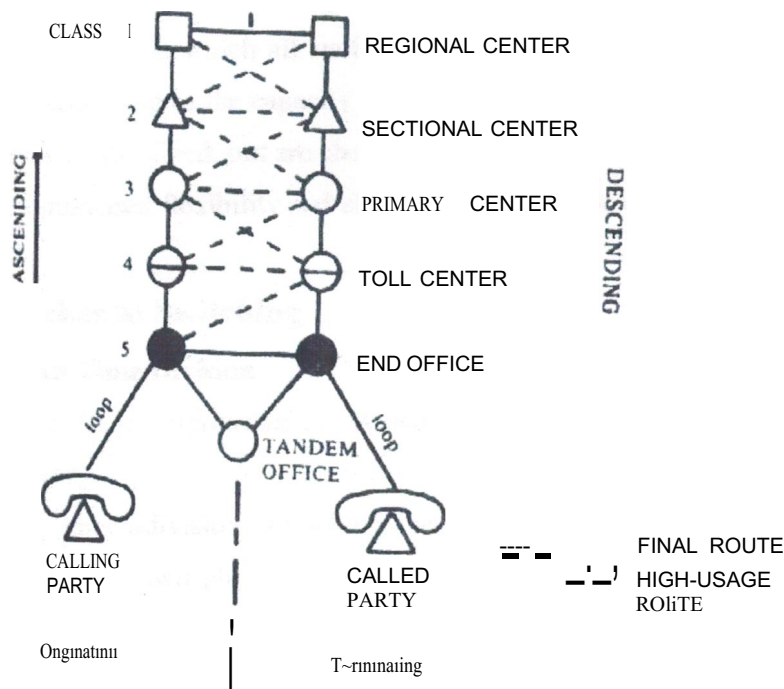


Figure 1.1

The number of offices increases as the tree is descended. There are only about 20 regional centers serving the United States and Canada, while there are well over 20,000 end offices.

Normally, a call will be connected using as few switching offices as possible. The intent is to keep the actual, final route low in the switching network. The preferred route is to connect across from an originating office to a lower terminating office using routes called high-usage routes. If this route is busy, then the next preferred route is to connect across to a center at the same level, or, if this route is busy, then across to the next higher center. If all the routes across are busy, then the route of last resort is to ascend higher in the originating office to the next higher office and attempt

do cross over again. The routes connecting centers in the originating offices and in the terminating offices are called inter-toll trunks.

The call proceeds through the hierarchy in the calling-office sequence and then descends the hierarchy in the calling-office sequence. Actual traffic continues at the time of a call determine the final route.

The routing used in the AT&T long-distance network is moving toward a dynamic non-hierarchical system in which all switching centers are equal and the route for each call is chosen according to the capacity that is available. Thus, with this newer system, the final routes are not fixed, but are chosen dynamically at the moment of the call. This system offers increased flexibility and efficiency in the use of the network.

1.6 Approaches to Switching

1.6.1 Space and Time Division

There are two basic approaches to switching: space-division switching and time-division switching.

In space-division switching, as Figure 1-12 shows, each telephone conversation has its own physical path through the switch. Each path is dedicated to that conversation only, and the physical connection is maintained throughout the duration of the conversation.

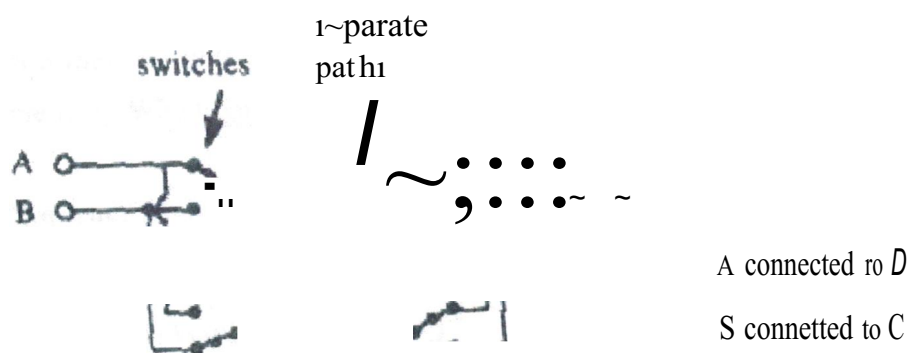


Figure 1.12

Time Division

In time-division switching, as shown by figure 1.13, paths for separate

conversations are separated in time, and connections between various paths are made for very short time intervals. In effect, each conversation is broken into samples, and these sample values are routed to their appropriate destinations along the shared path. This type of switching is most appropriate for signals that have been sampled and digitized.

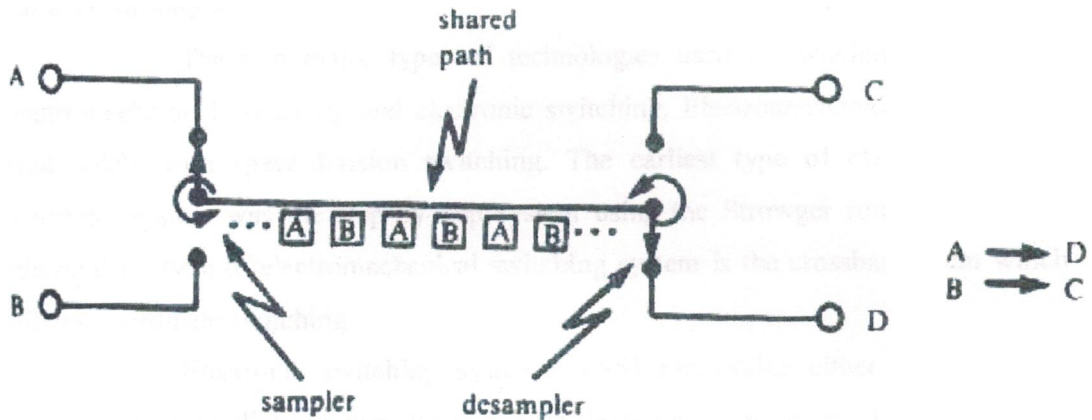


Figure 1-13

With space-division switching, signal paths are switched in physical space. With time-division switching, sample values of a number of signals sharing a common medium are reorganized, or switched, in their time sequence. The earliest automatic switches transferred physical paths and were space-division switches. The newest switches are called digital switches because they switch digitized signals by using a combination of space-division and time-division switching.

There is a third possible approach to switching, namely, frequency-division switching (see figure 1.14). With frequency-division switching,

1.6.3 Frequency Division:

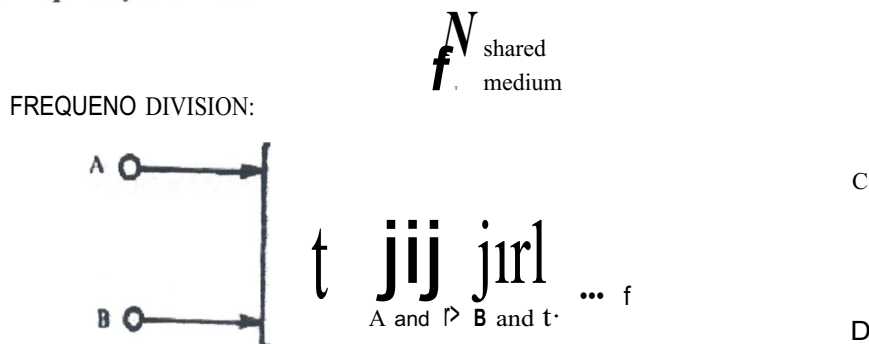


Figure 1.14

Bands would be assigned to different connections. This type of switching could be particularly applicable to the single broadband medium that might be shared by a number of users, such as the coaxial cable in a CATV system. Each pair of users would need to agree on a specific band of frequencies to be used for their particular conversations.

1.6.4 Technologies

The two major types of technologies used in switching systems are electromechanical switching and electronic switching. Electromechanical switching is used solely with space-division switching. The earliest type of electromechanical switching system was the step-by-step system using the Strowger rotary switch. An intermediate type of electromechanical switching system is the crossbar system which utilizes coordinate switching.

Electronic switching systems (ESS) can utilize either space-division switching or time-division switching. A coordinate switching network of small reed relays was used in the first electronic switching system, along with store program control. The newest electronic switching systems are digital switches using combinations of space-division and time-division switching. All switching is accomplished by using solid-state devices. These systems are described in more detail in the following sections.

1.6.5 Space-Division Switching

Normally, not everyone will want to converse with everyone else at the same time. Hence, it is not necessary to design a switching system so that all lines can be connected simultaneously with all other lines. The incoming lines can therefore be concentrated and distributed through a smaller number of switching paths before being expanded at the last stages of the switching process, as illustrated by the flow chart of Figure 1-15. In this fashion, the switching is accomplished in stages consisting of concentration, distribution, and expansion.

Elementary Switching Stages:



Figure 1.15

Clearly, it is possible that some paths in the switching system might become completely congested. In this case, some calls desiring service will be blocked. The blocking can occur at any stage, and the switching system must be designed to minimize blocking for the traffic that it handles. The actual design depends on the type of customer served by the switching office, insofar as different customers have different traffic patterns. The two approaches to space-division switching depend on the type of basic switch that is used at each switching stage. The earliest switching systems used the Strowger switch in which a set, or bank, of contacts was swept by one wiper contact.

The electromechanical switch invented by Strowger was a stepped rotary switch that could move in two dimensions (see Figure 1-16). The actual electrical connections were made by contacts that wiped across each other and were, therefore, subject to considerable wear and tear. The large amount of mechanical motion meant that a fair amount of time was needed to make the actual, final connection. The Strowger type of switch made a connection between one contact to one of many contacts.

Rotary Switching Network;

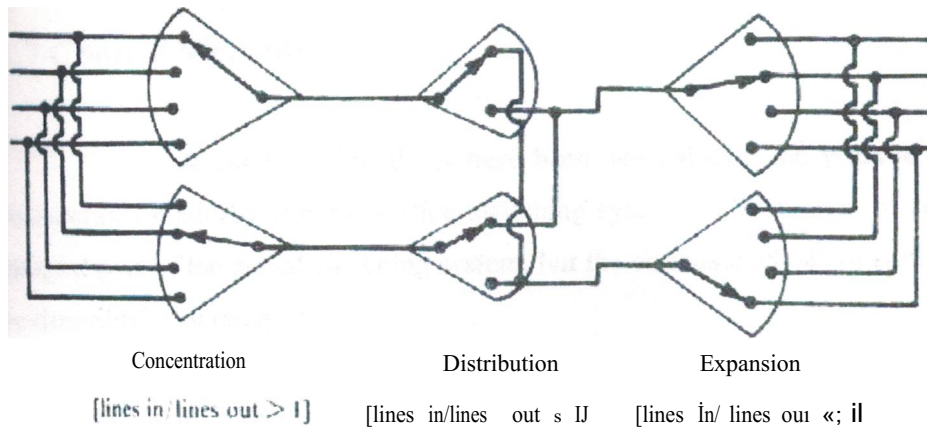


Figure 1.16

An improved approach to switching compared to rotary switching is coordinate, or matrix, switching (see Figure 1-17). In coordinate switching, connections are made at single contact points in a matrix consisting of all input lines and all output lines. The electrical connection at the contact point can be made by using a variety of technologies, such as conventional switch contacts, small reed switch contacts sealed in glass, and diodes and transistors biased to conduct or not to conduct.

Coordinate Switching Network:

Many telephone calls are to other telephone lines served by the same switching office. In these cases, the output lines from the switching system are connected to the incoming lines to make the actual connection. Such lines used for interoffice calls are called interoffice trunks.

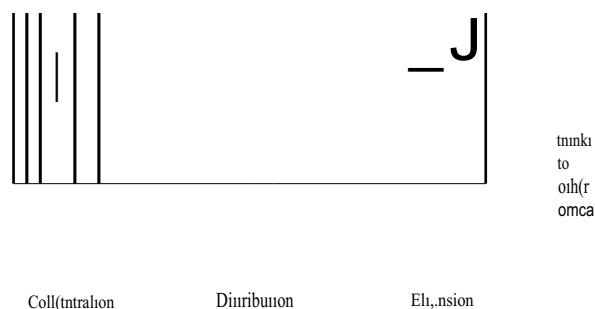


Figure 1.17

1.7 Control Methods

A number of methods have been devised over the years to control the connections made by communication switching systems. The method of control is an integral part of the actual switching system, but the various methods of control can also be described separately.

1.7.1 Direct Progressive Control

The oldest method of control is direct progressive control. With this method, sketched in Figure 1-18, each stage of switching responds directly to the digits dialed by the calling telephone. As one stage completes its connection, the next stage responds progressively to the dialed digits. The final connection through the switching system is made gradually as each stage receives its digits and progressively sets up the final path through the system.

The dialed digits contain the information necessary to operate each switch. The digit information is transmitted over the same lines that will carry the speech signal after the last connection has been made. The control path thus is the same as the speech path. Each switch has its own control associated with it.

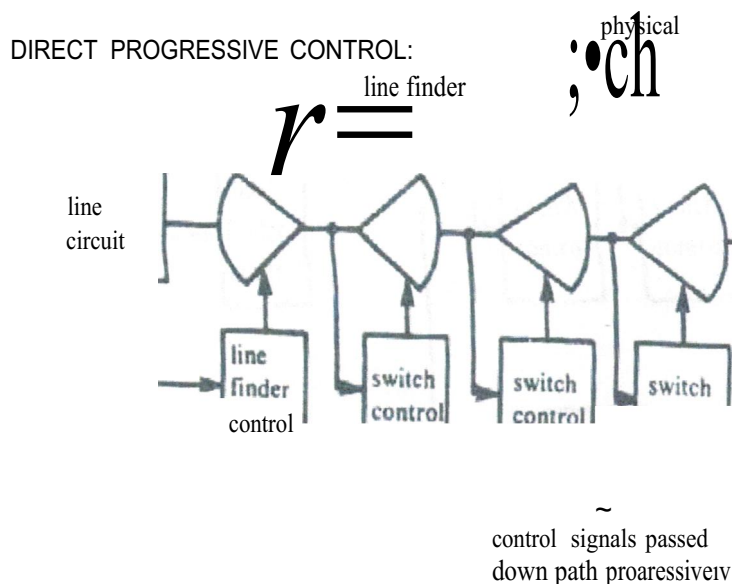


Figure 1.18

Direct progressive control is simple and economical. However, because the paths

1.7.2 Register Progressive Control

line
calling
line

line
finder
control

switch
control

switch
centre

REGISTER

TRANSLATION

21

Direct progressive control and register progressive control are used in step-by-step switching systems. The basic switch in step-by-step switching systems is the Strowger switch.

1.8 Step-By-Step Switching System

Invention

The step-by-step switching system was invented by Almon B. Strowger, an undertaker who was upset that the business in his town went to his competitor who's wife was the operator of the local telephone exchange. The basis of his invention was an automatic switch. It consisted of vertical and horizontal under the control of digits dialed at the telephone instrument by the telephone user. The first installation of the automatic switch was in 1892. A few years later, in 1896, the telephone dial was invented by some of Strowger's associates. The Strowger switch system was manufactured by the Automatic Electric Company and sold to the Bell, independent telephone companies. The first installation of a Strowger switching system in the Bell system took place in 1891.

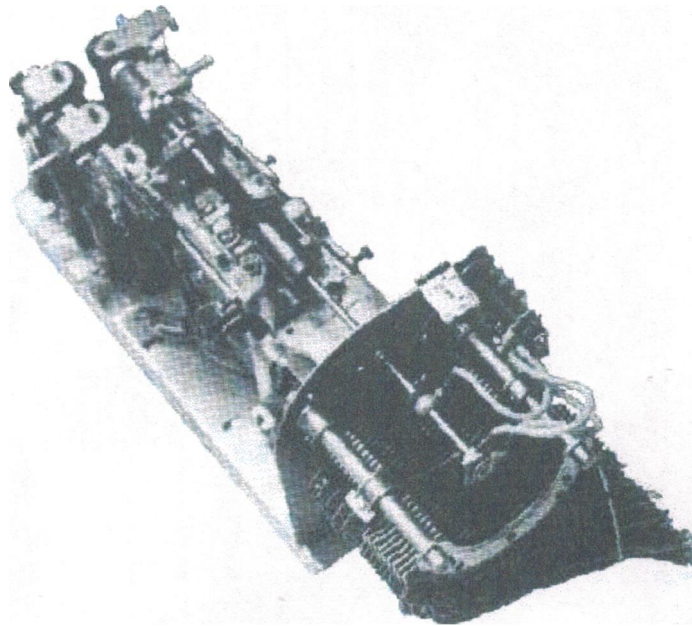


Figure 1.20

Photo of a Strowger switch, Note: The contact banks are shown along with the wiper arms and the relays at the top control the movement of the wiper arms.

The Strowger Switch The Strowger switch, shown in the photograph of Figure 1-20 is an ingenious, electromechanical device consisting of electromagnets and ratchets.

1.9 Panel Switching System.

The panel switching system was devised by the Bell System as its answer to the Strowger switching system. The first installation of a panel switching system was made in 1921.

The basic switch in the panel system was a Hoffmann affair, able to access any of five 500 terminals, as compared with access to only 240 terminals for the Strowger switch. This was accomplished by rotating cork rollers that moved the selecting magnets horizontally and vertically. The basic switch was extremely noisy, and the system needed considerable maintenance. Hence, there is virtually no panel equipment in service in the United States.

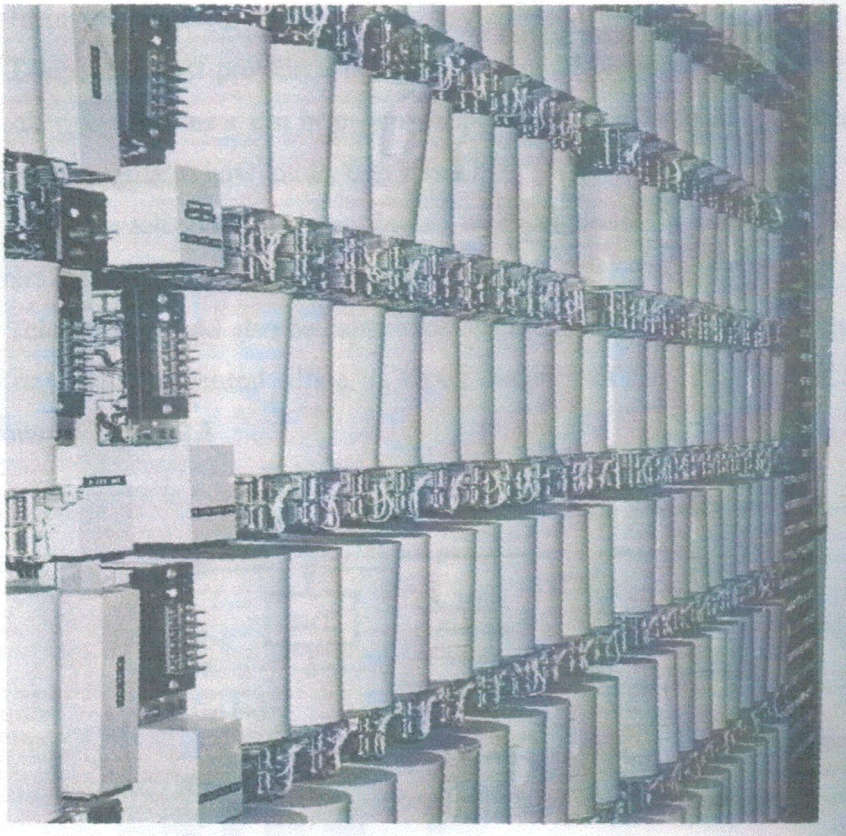


Figure 1.21

2. TELEPHONE & TELEPHONE NETWORK

2.1 Telephone Set

2.1.1 Basic Functions Of Telephone Set

1. Telephone must notify the user of an incoming call through an audio tone such as a ring.
2. Telephone must convert caller's speech to electrical signals. Otherwise, electrical signals must be converted to speech signals.
3. Calling method (subscriber numbers) may be pulse or tone.
4. Telephone must regulate the speech amplitude of the calling party by compensating for the varying distances to the local Telephone Company (Central office):
5. Telephone must gain the attention of the central office when a user requests service by lifting the handset.
6. Telephone must provide a nominal amount of feedback from its microphone to its speaker so that a user can hear person speaking. This feedback is called side tone. The side tone regulates how loudly one speaks.
7. When the telephone set is out of use, a pen-circuit must be provided to the central office.
8. Telephone should also be capable of receiving call progress tones (busy, ringing, and so on) from the central office. A block diagram of the conventional telephone set is shown in Figure 2.1



Figure 2.1

When preparing for answering a call, the telephone is lifted off of its cradle, and the on-hook switch is moved to the off-hook position. Power of the telephone set is derived from a -48V dc at the central office (see Figure 2.2). The power is delivered to the telephone set via the subscriber loop. Since most subscriber loops are two-wire pairs, a hybrid circuit is necessary to transform the two-wire transmission line into four wires. To compensate for the varying lengths of wire between the central office and its

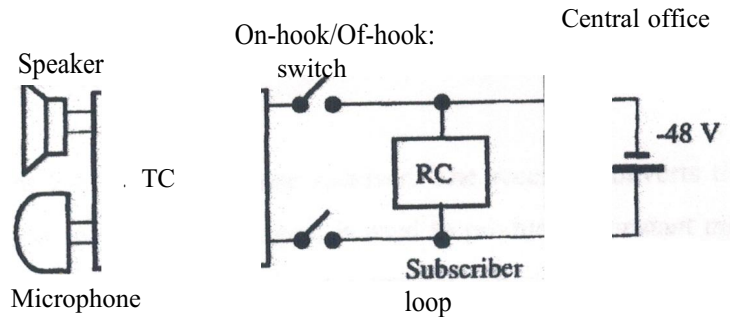


Figure 2.2

1.1.2 The Basic Elements Of The Telephone Set

2.1.2.1 a Transmitter

The part of the telephone into which a person talks is called the transmitter. It converts speech signals into an electric current that can be transmitted through the transmission system to the receiver. The most common telephone transmitter has used today is in principle like the one invented about 100 years ago by Thomas A. Edison's carbon microphone. Figure 2.3 illustrates a typical sectional view of a carbon microphone.

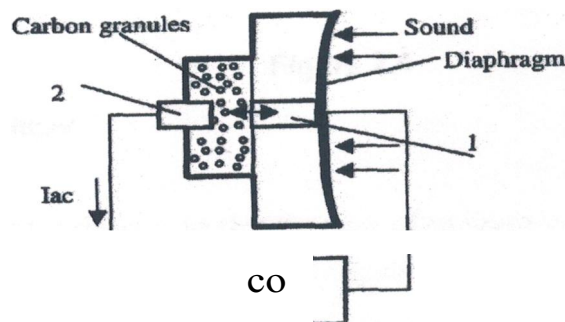


Figure 2.3

DC current provided by the Central Office (CO) is passed through two electrodes separated by thousands of carbon granules. One electrode (J) is attached to a diaphragm that vibrates, in response to the acoustical pressure of sound. The opposite second electrode (2J) is fixed. Vibration of the diaphragm causes the contact resistance between the two electrodes to vary inversely with pressure. As the resistance varies, the current (i_{ac}) varies inversely, thereby translating the acoustical message into the electrical signal that is transmitted to the central office.

2.1.2.b Telephone Receiver

Figure 2.4 shows telephone receiver. The receiver converts the ac current, i_{ac} back to sound. A permanent magnet is used to produce a constant magnetic field Φ_c . Insulated wire is wound around the armature, which passes the ac signal i_{ac} . The varying electrical current representing speech produces a varying electromagnetic field Φ_e . It alternately aids and opposes the permanent magnetic field; this combination creates an alternating total magnetic field acting on the diaphragm through armature. This causes the diaphragm to vibrate in step with the varying current and "moves the air" to reproduce the original speech signal.

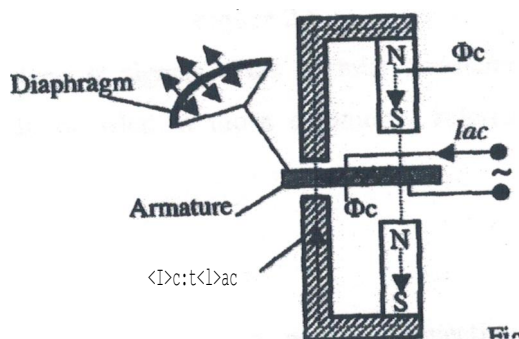


Figure.2.4

2.1.2.c Telephone Ringer

The function of the ringer is to alert the party of an incoming call. The audio tone generated by the ringer must be loud enough for the party to hear from a distance. Several variations of ringer are used in today's telephone sets. The most popular types are the conventional electromechanical and more recently, semiconductor sound

generators. In the United States, telephone companies will ring the called party with an a.c. ringing signal, typically 90 V rms, at 20 Hz. The ringing signal is superimposed upon the existing -48 V d.c. signals.

2.1.2.1 Telephone Hybrid

The telephone hybrid is used to interface the transmitter and receiver. The hybrid facilitates the use of a single-winding transformer in a manner to electrically separate the

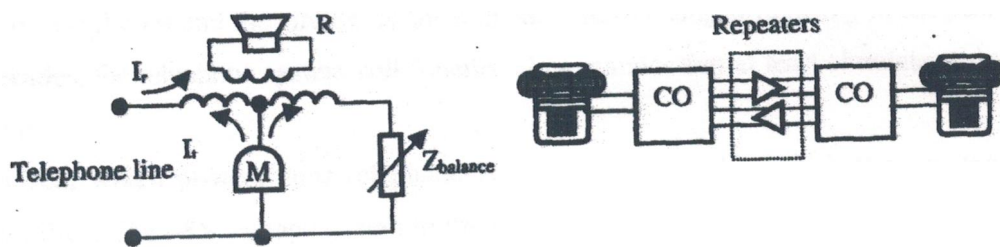


Figure 2.5

transmitted and received I_r signals. This permits simultaneous transmission and reception of the speech, or what is more commonly referred to as a full-duplex operation.

2.2 Telephone Network

First of all, I must officially advise against connecting anything other to the telephone line than equipment approved for the purpose by the telephone company or some other regulatory body. Telephone companies tend to be very strict about unauthorized gear hanging on their lines, and if something does go wrong with your gadget (like putting dangerous voltage to telephone line) you will be in deep trouble.

2.2.1 How Telephone Works

A telephone uses an electric current to convey sound information from your

me to that of a friend. When the two of you are talking on the telephone, the telephone company is sending a steady electric current through your telephones. The two telephones, yours and that of your friend, are sharing this steady current. But as you talk into your telephone's microphone, the current that your telephone draws from the telephone company fluctuates up and down. These fluctuations are directly related to the air pressure fluctuations that are the sound of your voice at the microphone. Because the telephones are sharing the total current, any change in the current through your telephone causes a change in the current through your friend's telephone. Thus as you talk, the current through your friend's telephone fluctuates. A speaker in that telephone responds to these current fluctuations by compressing and rarefying the air. The resulting air pressure fluctuations reproduce the sound of your voice. Although the structure of telephones and the circuits connecting them have changed radically in the past few decades, the telephone system still functions in a manner that at least simulates this behavior.

The current which powers your telephone is generated from the 48V battery in the central office. The 48V voltage is sent to the telephone line through some resistors and inductors (typically there is 2000 to 4000 ohms in series with the 48V power source).

The old

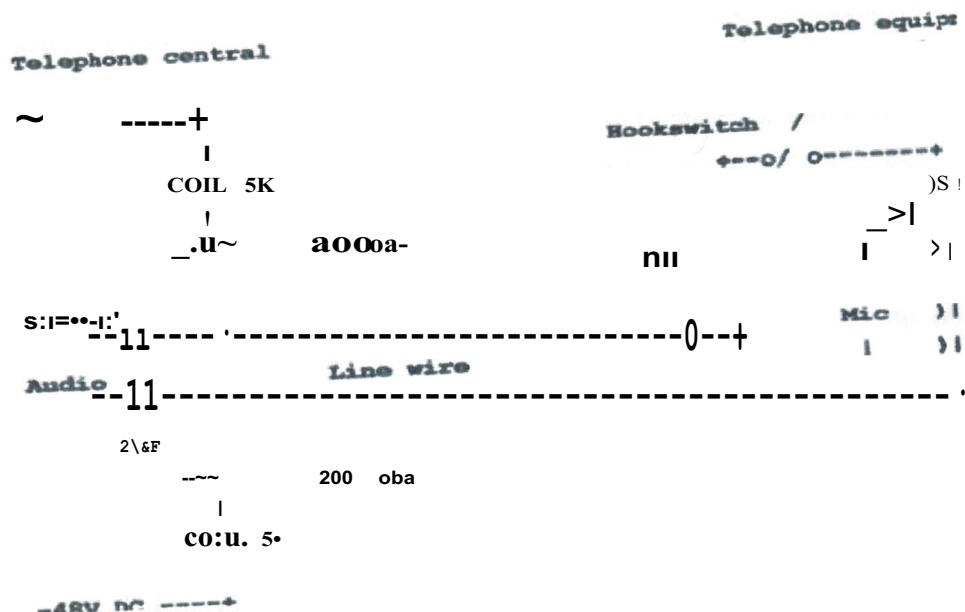


Figure 2.6

u go off hook, and current is dra

3v. This means that there is about

voltage between the wires going to telephone in normal operation condition. The resistance of typical telephone equipment is in 200-300 ohm range and current flow through the telephone is in 20-50 mA range.

2.2.2 Why 48V voltage is used in telephone systems?

The -48V voltage was selected because it was enough to get through kilo me s of thin telephone wire and still low enough to be safe (electrical safety regulation: n many

Countries consider DC voltages lower than 50V to be safe low voltage circuits). 4 V voltage is also easy to generate from normal lead acid batteries (4 x 12V car batterie n series). Batteries are needed in telephone central to make sure that it operates also w n mains voltage is cut and they also give very stable output voltage which is needed r reliable operation of all the circuit in the central office. Typically the CO actually i s off of the battery chargers with the batteries in parallel getting a floating charge.

The line feeding voltage was selected to be negative to make the electrochem il reactions on the wet telephone wiring to be less harmful. When the wires are at nega e potential compared to the ground the metal ions go from the ground to the wire insi d of the situation where positive voltage would cause metal from the wire to leave wh h causes quick corrosion-Some countries use other voltages in typically 36V to 6 V range. PBXes may use as low as 24 Volts and can possibly use positive feeding volt ge instead of the negative one used in normal telephone network- Positive voltage is m re commonly used in many electronics circuits, so it is easier to generate and electrol is in telecommunications wiring is not a problem in typical environment inside of e buildings. Some older offices employ battery reversal (swap DC feed to tip and ring o signal off-hook at the remote end.

2.2.3 What is signaling Current?

The current sent to telephone line as an another advantage besides that it supplies the operating power for your telephone. Telephone practice uses (or did use) twisted splices. These splices did not always make good connections. Placing a small DC bias on a long transmission pair is often done by telecommunication carriers to reduce poor connections, and noisy lines. The DC bias is often referred to as a "sealing current". So putting DC current through the cable sealed the connection and so improved the transmission.

2.2.4 Why Full Duplex Operation in Single Wire Pair?

Full-Duplex is a term used to describe a communications channel which is capable of both receiving and sending information simultaneously. Telephone sets (ordinary analog ones) have only 2 wires, which carry both speaker and microphone signals. The signal path between two telephones, involving a call other than a local one, requires amplification using a 4-wire circuit. The cost and cabling required ruled out the idea of running a 4-wire circuit out to the subscribers' premises from the local exchange and an alternative solution had to be found. Hence, the 4-wire trunk circuits were converted to 2-wire local cabling, using a device called a "hybrid".

This function can send and receive audio signals at the same time is accomplished by designing the system so that there is a well balanced circuit in both ends of the wire which are capable of separating incoming audio from outgoing signal. This function is done by telephone hybrid circuit contained in the network interface of the telephone.

2.2.5 What is the bandwidth of the telephone line?

A POTS line (in the US and Europe) has a bandwidth of 3kHz. A normal POTS line can transfer the frequencies between 400 Hz and 3400 Hz. The frequency response is limited by the telephone transmission system (the actual wire from central office to your wall can usually do much more). Nowadays POTS is sharply band limited due to the fact that the line almost always is digitally sampled at 8 kHz at some point in the circuit. The absolute, theoretical limit (with perfect filters) is therefore 4 kHz - but this

isn't reality, 3.4 kHz maximum frequency.

The low-bass frequency response is limited because of the limitations in telephone system components: transformers and capacitors, can be smaller if they don't have to deal with lowest frequencies. Other reason to drop out the lowest frequencies is to keep the possibly strong mains frequency (50 or 60 Hz and its harmonics) humming away from the audio signal you wish to hear.

2.3 Network Interface Details

The telephone has a circuit called network interface (also called voice network or telephone hybrid) which connects the microphone and speaker to the telephone line. Network interface circuitry is designed so that it sends only the current changes-the other telephone causes to the speaker. The current changes which the telephone's own microphone generates are not sent to the speaker. All this is accomplished using quite ingenious transformer circuitry. In theory the hybrid circuit can separate an incoming sound from the audio sent out at the same time if all the impedances in the circuitry (hybrids on both ends and the wire impedance in between) are well matched. Unfortunately, the hybrid is by its very nature a "leaky" device. As voice signals pass from the 4-wire to the 2-wire portion of the network, the higher energy level in the 4-wire section is also reflected back on itself, creating the echoed speech. The reason circuit does not work perfectly and you can still hear some of your own voice in the speaker.

The actual amount of signal, which is reflected back depends on how well the balance circuit of the hybrid matches the 2-wire line. In the vast majority of cases, the match is quite poor; resulting in a considerable level of signals being reflected back;

The signal which is reflected back is not always bad and in normal telephone some of it is really intentional by the design. The separation of the received and transmitted audio could be done much better with modern electronics than with old phones but people who use the telephone prefer to hear some of their own voice" back. Radio Shack's "Understanding Telephone Electronics" (copyrighted around 1985 I think) says this effect side us and gives the impression that this was indeed inevitable in order for the speaker to determine how loud they were speaking with reference to the called party.

2.4 Signaling

2.4.1 Manual-Exchange Signaling

In the early days of telephony, exchange service was accomplished with manual switching by a human operator. The telephone subscriber desiring service first had to alert the operator. This was done by turning the crank on the telephone, which caused a lamp to flash on the panel at the exchange office. The operator would see the flashing lamp and then plug in on that line. The calling party would verbally request the operator to make the connection to the called party.

The operator would then visually check the cords and jacks to determine whether a connection could be made to the called party. If not, the operator would inform the calling party that the called party's line was in use. If the called party's line were available, the operator would make a connection and inform the called party. The lamps for both the called party's line and the calling party's line would remain lit as long as the telephones were in use. As soon as one telephone was hung up, the corresponding lamp would go out, and the operator, noticing this, would unplug the connection.

If the call were from the local exchange to another exchange, the operator at the calling party's exchange would use special interexchange lines called *trunks* to reach the operator at the called party's exchange. The number to be called would be passed verbally from operator to operator, and the operator at the called party's local exchange would make the final connection.

The making of a telephone connection involved a large amount of human labor during the early days of telephony. Technology has, over the years, reduced and finally eliminated all human labor required in making a telephone connection. This was accomplished through electromechanical switching machines and various electrical signals to request service, forward telephone numbers, and set up the actual connection of the lines.

The general topic that deals with the various signals used to request service and to control the progress of the telephone call is known as signaling.

This latter function is supervisory in nature. Thus, the signaling on the subscriber loop; is used for alerting and supervision functions. Address information can be transmitted in two ways: on the subscriber loop. The flow of direct current can be interrupted by the telephone dial to generate dial pulses. These pulses are at a rate (if about 10) pulses per second. The second way that address information can be transmitted is in the form of unique two-tone combinations called touch-tone dialing.

Information is transmitted on the subscriber loop as either audible tones or recorded announcements. Four major tones are: dial tone, ring-back tone; line busy tone, and trunk-busy tone. Four generic frequencies (350, 440, 480, and 620 Hz) are used, either

Dial tone is a continuous tone formed by combining a 350 Hz sine wave with a 440 Hz sine wave by addition of the two waves. Ring-back or audible ringing is formed by the addition of a 440 Hz sine wave to a 480 Hz sine wave. The combination is called

There is one last alerting signal that is transmitted over the subscriber loop. This is the ringing signal which causes the called telephone to ring. It is a sine wave of 75 volts rms at a frequency of 20 Hz.

The oldest and most basic type of signaling between central offices is direct current, or dc; signaling. The presence or absence of a dc signal on a trunk would indicate whether the trunk was idle or in use. Normal or reverse direct current is cannot be transmitted over circuits derived from a carrier system, and hence, dc signaling cannot be used. The solution was to use a single-frequency tone, either in the voice band (200-3400 Hz), or outside the voice band (3700-3825 Hz). One popular

Unfortunately, there were problems with this in-band, single-frequency, signaling

scheme. For one, some people discovered they could generate their own tones for the fraudulent purpose of avoiding toll charges. For another, some speech signals could cause accidental disconnections. The use of an out-of-band frequency was not without its problems, too. One problem was the loss of usable bandwidth for the speech signal.

The address information was transmitted over the seized trunk by using two-frequency tones sent at a rate of 10 tones per second. The tones consisted of two-frequency combinations of 700, 900, 1100, 1300, 1500, and 1700 Hz. With the various combinations of these frequencies, it was possible to

Represent all ten digits and up to six control functions. This type of a signaling was called multiple frequency key pulsing (MFKP).

2.4.5 Common Channel Interoffice Signaling

In 1976, a new interoffice signaling scheme was first installed in the Bell System, and it is currently in use on practically all interoffice circuits. This new system is called common channel interoffice signaling, or CCIS for short.

With CCIS, as shown by Figure 2.7, a separate channel between the offices is dedicated as a data link for transmitting only signaling information. No signaling information is sent over the voice circuits. The switching machines used in most central offices are actually computers or processors that control the switching of voice circuits or paths. As such, the use of a data link to enable these computers to communicate with each other about the availability of the voice circuits between offices is quite consistent with the capabilities and requirements of the newer switching technology.

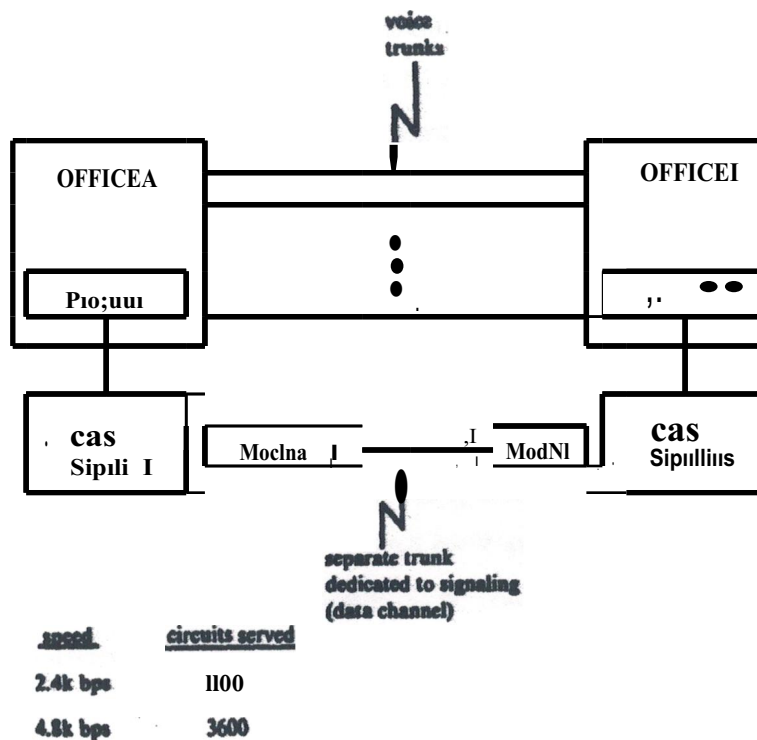


Figure 2.7

A single analog circuit is used to convey the digital signaling information. Conventional full-duplex modems operating at either 2.4 k bps or 4.8 k bps are used. Each signaling circuit can control about 1800 or 3600 voice circuits, respectively. Because the signaling information is transmitted over a separate circuit, there is a need to determine whether the transmission quality of the specific voice circuit is acceptable before it is connected for use. This is accomplished by performing a transmission quality check on each voice circuit before it is connected for service. The voice circuit is looped back onto itself, and a tone is transmitted down the circuit. The return level of the tone is checked to be certain that it is within specifications.

With conventional signaling, a busy tone is sent from the office closest to the called party, all the way back down the network to the calling party, thus tying up a full voice circuit. With CCIS, a busy tone is generated at the office closest to the calling party, hence freeing voice circuits for use with actual conversations.

CCIS at present does not transmit the calling party's identification to the terminating office. However, CCIS could be given the capability of doing so, which would make possible call screening at the local terminating central office. Also, it is

possible to envision that the calling party's identification might be transmitted all the way down the local loop so that the called party would know the identity of the calling party before answering the telephone. CCIS, therefore, might make possible many new services in the future.

2.4.6 Telephone Numbering Plan

During the early days of telephony, a total of 10,000 lines was the maximum number served by a telephone exchange. Thus, a four-digit number specified the party to be reached in an exchange. The exchange was specified by two alphabetic characters followed by a decimal digit, for example, WA5 for Waverly-five. Area codes were then introduced to specify the area in the country to be reached. Area codes are also called *numbering plan areas* (NP-A).

A special nomenclature is used to describe the telephone numbering plan. The symbol *N* is used for any of the decimal digits 2 through 9; the symbol *X* for any of the decimal digits 0 through 9; and *O* for the digits 0 or 1 only.

The standard format for telephone numbers in the United States has been *NO/ X-NXX-XXXX*. *NO/ X* specified the NPA; *NXX* gave the local exchange in the NPA; and *XXXX* denoted the specific subscriber line in the local exchange. Because the number of area-codes possible with the *NO/ X* format is being gradually exhausted, a new format of *NXX* is being introduced for the NPA.

The format *N 11* is used for special services. For example, 411 specifies directory assistance; 611 is the repair service; and 911 is for emergencies.

2.4.7 Local-Loop Signaling Design

The resistance of the local loop must not be too high; otherwise not enough current will flow in the line to activate the time relay at the central office. The resistance of the local loop depends on the total length of the loop and the gauge of the wire. Resistances for various gauges of wire are as follows:

microphone and one resistor.

.nnoqrao 'J~)Jn~ds 'l~uu,DJSUtIJ4 pμ.q,{qJD .pasisiroo JI.UX-P ~.:>!DA ...uo.q.cl; }1,a.1 JBUO!-!P~.JI
V 1ln~.ip ~;">fPA pm? !ln;">J!;, ~ll!~!P .1~füiμ JO SIS!SJ..ioa -:{Uoqd~t 1:tmliON

2.6.1 Simplified Traditional Network Interface

:)}Jnb iJursn p::n.isndmO(Y.)l? S! sp.n, HV "J;:l)fR~ds ~lll Ol pIY..ls 1ou ~.m S~llll~u~1r-uonidoJ:JJ~U
UA\0. s,~.uo.q.d~p.1 ;!Q}. .lP-!ll-i\> .sa2u1:ip 11a11r10 ,gn.1. 'Ja)J.B.g<ls ,~n Di .sosnao ,~UOI{OOE;l.J.;}ij-i-0
;;>q1 s;;>ut?q::, uiauno ';;)m 11UO spu~s 'il '1u:q1 os v~~!sgp 'S! k1!n::,r!a :;>3t?JJglll 11(JOMcl~N

26 gauge - 83 ohms per 1000 ft,
 24 gauge - 53 ohms per 1000 ft,
 22 gauge - 12 ohms per 1000 ft,
 19 gauge - 7 ohms per 1000 ft.

The maximum resistance of the local loop can be calculated as follows. The telephone instrument requires about 23 mA for the carbon transmitter to operate reliably. The common battery at the central office has an electromotive force of 48 volts. Thus, the total resistance of the circuit must not exceed $48/0.023 = 2100$ ohms. The resistance of the telephone instrument is equivalent to 400 ohms. Similarly, the resistance of the central office unit is also 400 ohms. Hence, the resistance of the loop must not exceed $2100 - 800 = 1300$ ohms.

The maximum loop resistance of 1300 ohms, determines the wire gauge for a given loop length.

2.5 Telephone Line Parameters

Telephone line resistance, capacitance, and inductance do not depend on the voltage or current on the line.

2.5.1 Line Balance

For telephone local loops, crosstalk is related to how well balanced the circuit is. Loop current does not affect that balance, even if excessively high. If the balance is not good enough, you can hear crosstalk from other telephone lines or from other noise sources. The balance of the telephone line is determined by the circuits connected to telephone line ends (typically line transformers) and the quality of the telephone cable (wet cable causes transient balance problems if wires are in contact with the water).

2.5.2 Loop current effects

The detrimental effects of excessive loop current would be distortion caused by saturation of transformers ("repeat coils" in the vernacular). Within the range of acceptable loop current (up to 120 mA), no transformer used in telephone equipment should become saturated. If an inferior transformer is used, or if loop currents were

significantly higher than 20mA, then distortion could be expected. Neither situation is common,

2.6 Network Interface in Telephone

The telephone has a circuit called network interface (also called voice network or telephone hybrid) which connects the microphone and speaker of the telephone line. Network interface circuitry is designed so that it sends only the current changes the other telephone causes to the speaker. The current changes which the telephone's own microphone generates are not sent to the speaker. All this is accomplished using quite ingenious transformer circuitry. The circuit does not work perfectly and you can still hear some of your own voice in the speaker (it could be done better nowadays but people who use the telephone prefer to hear *some of* their own voice back),

2.6.1 Simplified Traditional Network Interface

Normal telephone consists of ringer, dialing circuit and voice circuit. A traditional telephone voice circuit consisted of hybrid transformer, speaker, carbon microphone and one resistor,

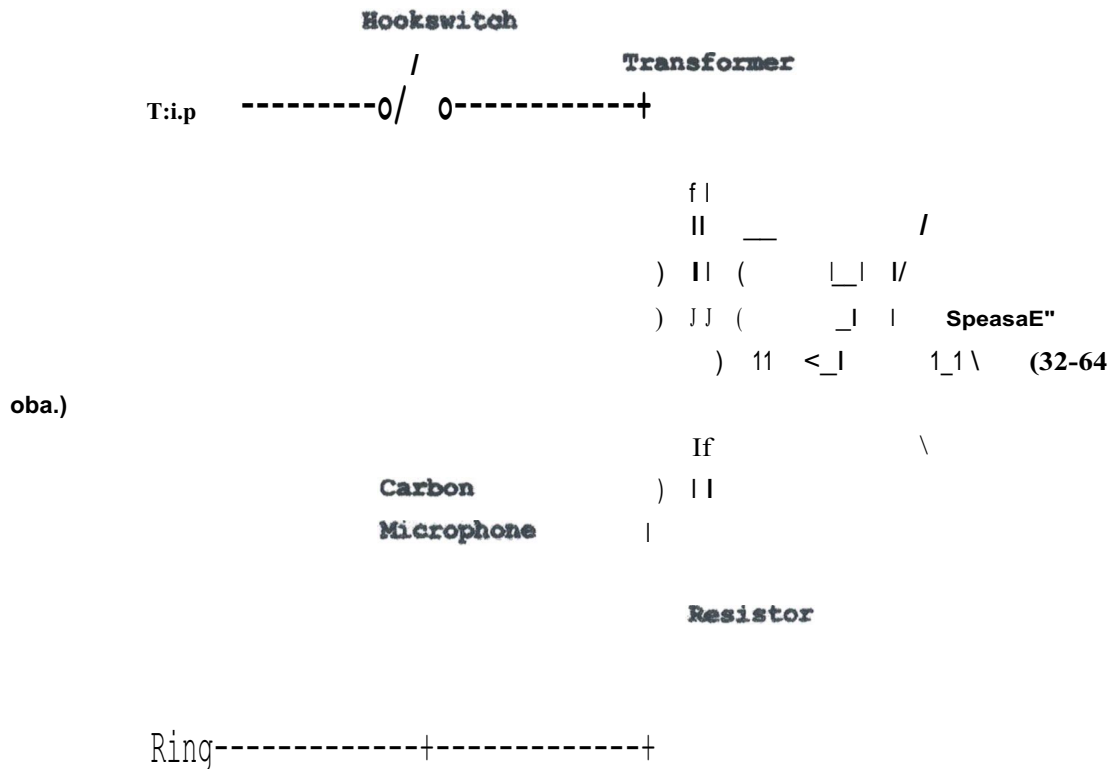


Figure 2.8

The circuit is designed; so that the impedance, at audio frequencies looks like about 600- ohms, The audio Impedance is controlled by the transformer characteristics; carbon microphone, speaker impedance and the resistor in series with the transformer.

The DC resistance consists of the transformer coil in series with the resistor and part of the coil in series with carbon microphone. The carbon microphone is put to the transformer so that the changes in the current flowing through it do not generate voltage to the secondary coil where the speaker is connected.

Modern telephone circuit is much more complicated because they typically include compensation for the attenuation caused by long subscriber lines. This compensation is done so that the audio levels are controlled according the current flowing through the telephone (longer line has more resistance so there is less current which you get from 48V source through it).

2.6.2 why carbon microphone in telephones ?

Carbon mikes were the first microphones and consisted of a small button of carbon powder connected to a metal diaphragm; When the sound pressure waves hit the diaphragm, the carbon grains changed their electrical resistance. When a voltage source is applied between the microphone wires a variable current is generated. This is how the first telephones were constructed, and many phones to this day still use the idea. Carbon microphones have poor frequency response and bad signal-to-noise ratios and they are only suitable for telephones and such communication applications.

2.6.3 Typical European Network

The following network circuit schematic was shown in BUILDING A11D
USING PHONE PAT-Cfl...ES by Julian Mecassey:

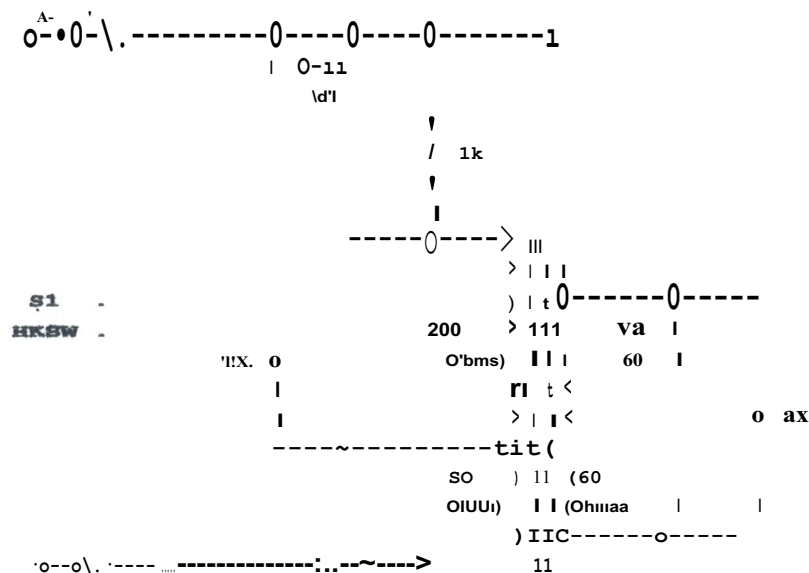


Figure 2.-9

Note: I have edited the schematic by replacing the component numbers with the component values listed in component list

2.6.4 Simplified U.S. Standard "425B"

This circuit is put here to show an example of the electronics inside typical

traditional telephone which uses hybrid transformer circuit. Modern telephones usually have special ICs to do the same things without the transformer. The circuit tries just to be an example what's an inside typical old telephone for those who want to know how telephone works. Building this circuit is only a good idea because the circuit diagrams does not have all component values and the circuit is optimized only for telephones (it is not good for anything else).

This circuit is taken from UNDERSTANDING TELEPHONES article by Julian Macassey. Component values may vary between manufacturers. The circuit is designed to operate with standard telephone speaker (RX) and carbon microphone (TX). Connections for, Dials, Ringers etc. not shown to keep the picture a little bit clearer. The circuit is quite complicated because it is optimized for use in standard telephone which is used in various conditions. Varistors VR1 and VR2 are used for loop compensation circuit which tries to keep the telephone volumes (incoming and outgoing) at suitable levels even if the local loop attenuation varies. This compensation can be done because longer local loop which has more attenuation has also more resistance, so less current passes through the telephone. If the loop is very short there is more current passing through the telephone and the varistors cause more signal attenuation inside the telephone hybrid.

The hybrid circuits in telephone sets are deliberately mismatched, so that you can hear yourself in the earpiece when you speak. This is called "side tone".

and GN respectively. Black and Red are mike leads BD they connect to B and R
~ti.v:tly.

Ring. er: Conneenhe single winding, M series with the; A-K. Capacitor field th,,
whole thing across the line. Rotary dial: Blue and Green go to interrupter (~t E atid:
RR) Touch-tone dial; Green is + line in and connects to net F. Black is + line out and
~f{f}f{f}S t-o 'f'et. RR. Qtg, 'Blk 'is - 'n{f,l'e aim sacntl -oo~.ct~ t-o lJet £. ~e1MGirn'lB t}nput
common and connects to net R. Blue is signal output and, C<, nme-cis-lo- Ret B.

Hook switch: You'll find many variants of this in different units; some configurations
switch both sides of the line; some only one; some switch out the ringer when off-hook. One
, Switch switches the connection between LI and C. Another switch switches the connection
between LI and RR. Lr-ne-itr. 'Green and Red: connect to LJ and L'. L'ry use *perfectly*. If the
touchtone dial doesn't work then flip it Tenr.

2.7 Telecom Hybrid Circuits For Other Equipment

2.7.1 Traditional transformer hybrid circuit

The transformer type was the most used to make telephone hybrids (around 1904
'Or:tro}, was fin) in winding "trails, future. 'f.w0.,,0f'l.tloo wet>e~lteenm, for..oun,h;ynriasoircuit.

Richard Harrison gave me the following description how to make such hybrid
-cw.mit: To make the hybrid, strtp two coils together in each transformer (series-aiding
in each case). Call them primaries. One primary will serve as the 4-wire transmit
, cormoofron. The other, prim<y-WM -sew:e-as, the -4-, wire receive, connection. If aur, coils,
two on each transformer remain undedicated at this point. Connect the start terminal of a
secondary coil on one transformer to the finish terminal of a. Hke. eoH on the other
transformer. The other two terminals of this pair of secondary coils will be dedicated to
-a-bainm;-iflg~net-wer<.

Two coils now have no connections; yet Connect the start terminal of the coil
on one transformer to the start terminal of the coil on the other transformer. The other
two terminals of this pair of secondary coils will be dedicated to the 2-wire line.

„Suame there is :.Fp{}larity, re<Ver-sal-in the -inte~nne&ti@n :i:n, one of the two paths
between the two transformers, no coupling will exist between the transmit and receive
connections of the 4-wire paths (provided perfect balance in the line-balance network

again, the 2-wire line). The 2-wire line will, however, be coupled with the transmit and receive pairs. Since the 4-wire line, the hybrid is supposed to do.

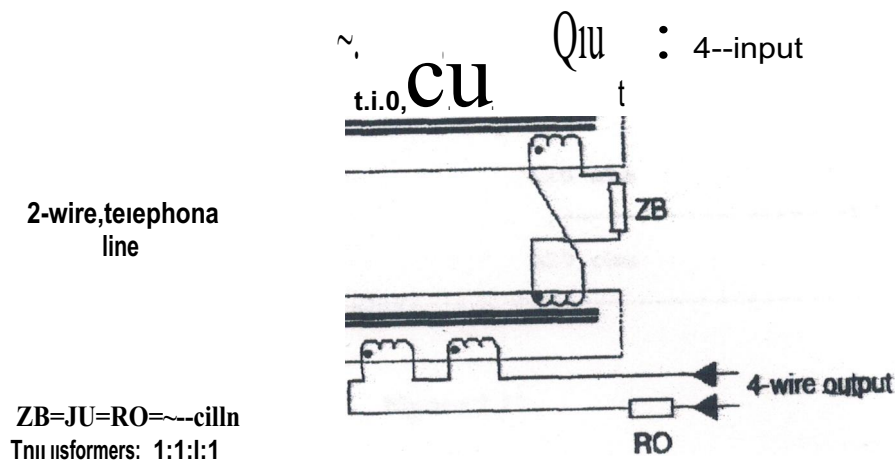


Figure 2.11

The advantages of the traditional circuit are: High isolation - No dc path exists between any lines. The circuit is completely passive and precise balance can produce almost any desired transmission loss. You should get very good results when you implement this circuit using high quality audio transformers (for example: the Lundahl Western Electric HI-6 "repeat coils". Lundahl Transformers Hybrid Transformers etc.).

2.8 Siemens and ITT Resistive Hybrids

This is a simplified circuit diagram you can make a simple 600 ohm hybrid as such. The circuit is indeed a Wheatstone bridge consisting of four 6-20 ohm impedances (one of them is telephone line in series with a DC blocking capacitor);

Here the wires marked with LINE .2W are the wires of the 2 wire duplex line. Wires marked with: RX and TX belong to 4 wire line. RX is the pair where the received audio from 2 wire line comes. TX is the pair where the audio which is to be transmitted to 2 wire line are sent. The component marker with ZZZZ models the telephone line impedance (typically around 600 ohms).

2.8.2 Connecting Hybrids and Telephone Equipments

Sometimes there is need to connect normal telephone equipments directly to the telephone hybrid circuit without any connection to public telephone network. This kind of interfacing is needed for example for telephone equipment measurements using hybrids or for interfacing telephones to computers through a hybrid circuit. There are few different ways to do the interconnection of hybrid and telephone equipment.

2.8.3 Simple Interconnection with no Power Provided to the Line

Simplest interconnection is just wiring the telephone equipment to the hybrid. This kind of simple interconnection works for cases where the telephone equipment does not need any telephone line loop current to operate (normal telephone averaged or loop current and can not be used in this way).



Figure 2.13

2.8.4 Simplest Powered Circuit

This circuit is suitable for simple telephone equipments like normal telephones connected to transformer based telephone hybrids which can withstand full telephone line OC current (use "wet" type transformer which can withstand at least 50 mA DC without saturation).

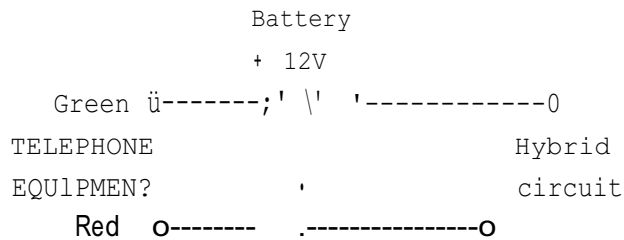


Figure 2.14

The circuit works so that the battery voltage limits the current to the telephone equipment. The current taken from the battery is limited by the resistance in the telephone itself and the DC resistance of the hybrid. If you fear of excessive current, you can put a 220 ohm 1 W resistor in series with the power supply. This will limit the current below 50 mA in all cases and does not cause too much impedance mismatch to the circuit.

2.8.5 General Hybrid interface

This is a general circuit suitable for interfacing "dry" hybrid circuits to practically any telephone equipments (works also for wet hybrids):

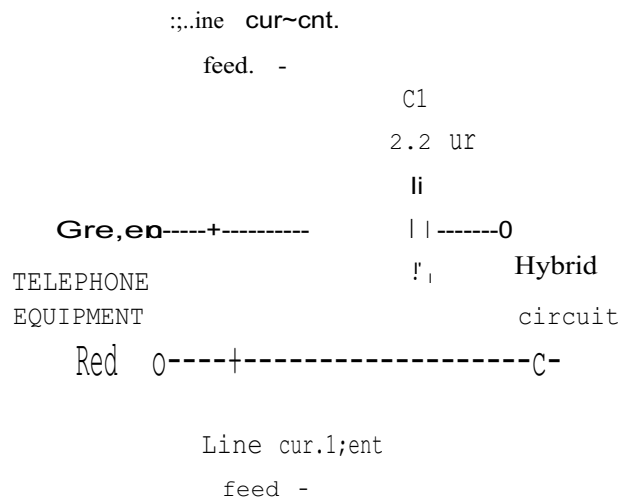


Figure 2.15

This circuit uses a capacitor C_1 to isolate the line current fed to the equipment from the hybrid circuit but still passes the audio signals. The capacitor should have a voltage rating so high that it can withstand the voltage which might be present in the line. The value of C_1 is not very critical, all values from 1 μ F to 50 μ F will work well. A "dry"

capacitor type like polypropylene is preferred capacitor type to be used.

The current feed is an external circuit which is used to supply the current to the telephone equipment in use. For normal telephone equipments and ideal current source with nominal current in 20-30 mA range and the open circuit voltage in range 12-48V would be ideal. NOTE: The source must be current source type. Normal voltage sources like batteries or normal DC Power Supply does not work for this because of their low internal impedance which would just short-circuit the audio.

If you do not have a suitable ideal current source, you can use other methods for making "close enough" substitute for telephone applications. The closest thing to a traditional power supplied by Telephone Company would be a 48V power source fed through around 1 k ohm resistor and 2H inductor. If you use lower resistance values you can use lower voltages. The 2H coil is needed to keep the impedance on audio frequencies high so that the "Short circuit" the signal does not cause serious impedance mismatches.

If the actual impedance matches are not very important, then you can try methods like 12V power source fed through the coil of a 12V relay or through 680 ohm 1W resistor. Both methods work in some cases, but can cause impedance mismatch which can cause poor operation of the hybrid (the isolation between incoming and outgoing audio signals will not be very good).

2.8.6 Components for Telephone Line Interfacing

If you are looking for components relays, and transformers, for making telephone interface; check the following companies:

- Clare
- Mitlton:
- Prem Magnetics
- BottrriS.

Using ready-made type approved interface can make designing small volume telecommunication product more easily, but unfortunately those ready made DAA are usually more expensive than the discrete components. The following companies make DAA products:

- X-ecom makes miniature telephone interface modules
- Cermetek has a selection of DAA products
- Siemens makes optically isolated DAA Modules: DAA-20.0&

The frequency response of the line depends on the line length. When line gets long high-tones drop-off much more quickly than the low tones (with the obvious effect on speech). It's not as difficult to tell the distance an analog phone is from a central office - {simulating an analog HLI is the connector}; it there aren't "highs" - it's far.

Take also note that the telephone equipment has a huge effect on the speech quality. For example carbon and electret handset microphones have radically different frequency responses. The frequency response and, overall sound quality of carbon microphones used in old telephones are not very good. Many modern telephones with electret microphones give better sound quality.

2.9 Telephone Line Details in Different Countries

Normal telephone line is theoretically designed to be 600 ohm resistive impedance. This 600 ohm is kept as international reference for designing telephone line equipment (typically the signal powers are measured to 600 ohm load). In practice the telephone line does not take pure 600-ohm resistance. The cable and equipments used by the telephone companies have effect what the real impedance is.

Telephone equipment which is designed to operate with 600 ohm loads will operate with these real-life lines but its performance is worse than in ideal situation. Typically the modems are designed for 600-ohm reference impedance because they can handle the side tone, but for best performance the telephones are designed to the exact line impedance.

When best performance is needed the circuit should be exactly matched to the impedance of the real telephone lines. Matching the hybrid circuit to the real line impedance (instead of 600 ohm) will improve the feedback typically by 3--6dB. 20dB side return is easy to achieve, but 3-0dB is also not too difficult provided you can measure the line impedance and take steps to build a correct balancing network.

Different countries have different characteristics on the telephone line parameters. Here are some impedance models for typical lines in different countries.

2.9.1 USA

Normal telephone subscriber lines in USA (0.4-0.6mm subscriber PE insulated vaseline filled cable) are 770 ohm resistor (with 2uF series capacitor) and 47nF parallel capacity.

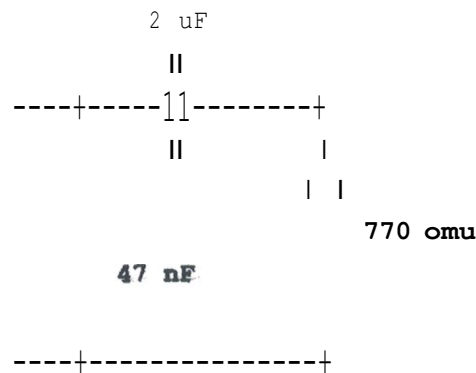


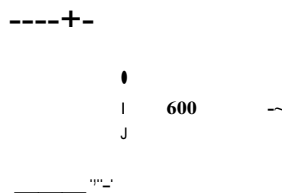
Figure 2.16

This diagram is referred to 800Hz, but impedance is rather complex, and varies from high value at low frequency and drops to ca. 150 ohm on 10kHz and 120-125 ohm above 100kHz.

Some telephone lines can have higher impedance (typically 1100 ohms in lines with ionatlingtons or telephone-air cables).

2.9.2 Finland

The equipments connected to public telephone network in Finland must meet NET4 (ETS 300 001) technical specs. All power specs and return loss measurements are taken so that the reference impedance is 600 ohm resistive.



The return loss of the terminal equipment must be greater than 10 dB when compared to 600-ohm reference. This measurement applies to telephones, modems and other terminal equipments. NET4 technical specs are European specs and they are used in many European countries (NET4 is actually a collection of different specs in use in different countries). Telecommunications Administration Centre in Finland also

mentions in its regulation THK 20 1/1997 M that the telephone line equipment can be measured against the 600 ohm resistance mentioned in NET4 or complex impedance of $Z = 270 + j50$ ohm (750 // 150 nF). Here is a picture of that complex reference impedance:

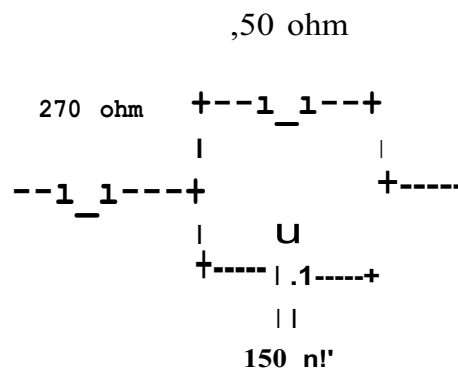


Figure 2.7



Typical cable used in subscriber lines has following characteristics: 0.5 mm diameter wire, loop resistance 182 ohm/km and pair capacitance 39 nF/km.

2.10 Notes about Telephone Transformers

Telephone line interfacing transformers are usually called 600:600 ohm transformers (or 1:1 ratio 600 ohm transformers). The both markings tell that the transformer has (around) same number of turns on both primary and secondary coils and they are optimized to operate at 600 ohm load. The 600 ohm load does not tell the primary or secondary coil resistances or impedance, it just tells in what kind of application the transformer is designed to be used. The DC resistance of typical telephone line transformer coils is around in 40--150 ohm ranges and inductance is typically in range of few henries:

A 600:600 transformer is optimised for 600 ohms use, but of course will work over a range of impedances more or less well (for example you lose a whole octave at the low frequency end if the impedance is 1200 ohms).

3. Comparisons of Communication Modalities and Media

3.1 Voice and Video

It is enlightening to compare various communication modalities in terms of their analog and digital requirements.

Voice telephony requires a baseband analog bandwidth of 4 kHz. The digitization of the speech signal results in the data rate of 64 kbps. This digitized signal can be compressed in its data rate by capitalizing on the many redundancies that occur in the speech signal. The degree of compression depends on how much degradation of quality is acceptable. The 64 kbps signal can be reduced to 3.2 kbps with virtually no noticeable degradation of quality. The digital signal can be further reduced to a data rate as low as 1.2 kbps, but the degradation of quality would be quite noticeable, and it sometimes might be difficult to recognize what was said.

A television video signal requires a baseband analog bandwidth of 4.5 MHz. The digitization of the video signal results in a data rate of 50 Mbps. There is a considerable amount of redundancy in video images, both within an individual frame and also between frames. The digitized video signal can be compressed to 1.5 Mbps with some degradation of quality, particularly for those portions of the image that move appreciably from frame to frame. Compression to the data rate as low as about 0.8 Mbps is even possible, but the degradation of quality would be more noticeable. The preceding can be summarized in the following table:

ANALOG	DIGITAL	
<u>Bandwidth</u>	<u>Baseband</u>	<u>Compressed</u>
4 kHz	64 kbps	1.2 kbps
4.5 MHz	50 Mbps	0.8-1.5 Mbps

Table 3.1

An interesting observation is that the ratio of the requirements for video compared with

speech is always about 1000 to 1 when like methods of encoding are considered, indeed, the old adage that "a picture is worth one-thousand words" appears; moreover well grounded and applicable even to communication technology!

3.2 Text and Image

Electronic mail holds great promise for the future as a new communication modality. There are two ways in which the information on a sheet of paper can be encoded and communicated electronically. In one way, the image on the sheet of paper, be it text or pictures, is scanned electronically and converted into an analog or digital signal. This signal can be transmitted to a distant location where the image of the page will be reconstructed. This type of image scanning and reconstruction is called facsimile transmission. The second way of encoding the information on a sheet of paper is in terms of the ASCII symbols that comprise the text (ASCII is the American Standard Code for Information Interchange.) Facsimile can cope with both text and pictures, but ASCII can only cope with text.

The digitalization of text and images depends upon several factors as: the resolution of the scanning process, the number of quantization levels required for each point, and the sophistication of the compression technique used to reduce redundancy. A resolution of 240 lines per inch is extremely good for most documents.

If a resolution of 200 lines per inch is used in the vertical direction with a corresponding resolution in the horizontal direction then an 8.5-inch page has a little less than 4 million picture elements. If a single bit is used to encode each picture element, then 4 million bits are needed to encode the image digitally.

There is a great deal of empty space (for most pages). Hence, a large amount of information is feasible for facsimile transmission. Compression, by 20 to 1 is possible, thereby reducing the number of bits per page to 200,000, or possibly even less, if fancier compression techniques are used.

A page of text is most efficiently encoded in terms of the ASCII characters, which go, moreover, to a full page of text consists of 60 rows of 80 characters each; then the full page would contain nearly 5000 characters. At eight bits per character, a full page of text would require 40,000 bits. There is redundancy in text because some combinations of characters occur more frequently than others; and thus the number of bits needed to encode the characters on a page of text might be reduced by at least a

factor of two to about 20,000 bits. Such reductions are rarely performed because ASCII is rarely as efficient compared with other communication modalities.

3.3 Text and Speech

A fairly good rate for reading aloud is about 120 words per minute. If an average word is six characters long; then the text equivalent of one minute of reading aloud would be 720 characters or 5760 bits using eight-bit ASCII encoding. The text equivalent of a five-minute telephone conversation would be 5 X 5760 = 28,800 bits. At 64 kbps, the five-minute telephone conversation would require 5 X 60 X 64,000 bits, or nearly 20 million bits. Clearly, text is considerably more efficient than speech in its use of bits by a factor of roughly 700 to 1.

Voice mail is a funneled, electronic mail system, wherein actual speech messages are stored digitally for later retrieval by the intended recipient. The digitization of speech at a 64 kbps rate would require a fairly large amount of storage for a short time, but the shortest messages. Hence, compression is used with most voice-mail systems. A nominally acceptable quality might be possible at a rate of 9.6 kbps.

For the sake of comparison, it will be assumed that the speech message is encoded at a rate of 11.2 kbps; although the quality now available at such a low rate would not be acceptable. A one-minute message would thus require a total of 60 X 11.2 k bits, or 72,000 bits. The textual equivalent, of the speech message would be 120,000 bits, which would require less than 6000 bits using eight-bit ASCII encoding. Thus, text is far more efficient than speech, even with maximum compression of the speech signal.

3.4 Total Traffic

The yearly total traffic in bit equivalents for various communication modalities and media can be estimated. The traffic represents the gigawatt capacity of the system to store all the messages for each medium.

The average daily number of local and toll telephone conversations for Bell and independent telephone companies is 80 million (Slat Abs: 1984). Assuming that the length of an average telephone conversation is three minutes and that the speech is encoded at 64 kbps; we have a total yearly traffic of 1.4×10^{18} bits, or 3400

quadrillion bits.

For television programming, it is assumed that there are three major networks, two minor networks, and seven cable broadcasters, for a total of 12 channels. Each channel will be assumed to broadcast 20 hours of programming each and every day of the year. At 50 Mbps digital encoding of the television signal, this amount of programming is equivalent to 1.6×10^{16} bits, or 16 quadrillion bits.

In 1982, 71.8 billion pieces of first-class and second-class mail were carried in the United States. Assuming that each piece of mail contained the equivalent of one full page of text (4800 characters), we have a total yearly traffic of 3×10^{15} bits, or 3 quadrillion bits.

It is estimated that about 3 billion checks will be negotiated in 1985 in the United States. Each check can be assumed to contain about 200 characters of text. Thus, the yearly digital equivalent capacity of all checks, assuming eight-bit ASCII encoding is used, is 6.1×10^{13} bits, or 0.06 quadrillion bits.

There are about 9200 newspapers in the United States, about 6800 of which are weeklies. The average length of a newspaper is about 100 pages. This means that about 123 million newspaper pages are published a year. It will be assumed that each page contains about 7000 characters of text, which is equivalent to 56,000 bits using eight-bit ASCII encoding. In addition, it will be assumed that each page contains about 200 square-inches of pictures and advertising. Using 200 lines per inch sampling and 2-to-1 image compression, this pictorial information is equivalent to 400,000 bits per page. Thus, each newspaper page is equivalent to about 456,000 bits. Thus, yearly production of newspaper publishing is 123 million times 456,000, which equals 56×10^{12} bits, or 0.06 quadrillion bits.

The preceding can be summarized in the following table: 3.2

<u>MEDIUM</u>	TOTAL YEARLY TRAFFIC
	(in bits) (in billions)
Voice Telephony	3400
Television Programming	16
Mail (first-class and second-class)	3
Checks	0.06
Newspapers	0.06

Table 3.2

Clearly, voice telephone traffic swamps all other communication modalities and media. This is not surprising because each telephone conversation is a unique message, requiring a considerable amount of bits.

3.5 Philips Recording Interface

The first circuit is Philips LFH0117/00 telephone recording adapter, which is not manufactured anymore I think. The circuit is quite typical telephone recording adapter design. I have used successfully for getting audio from telephone line to soundcard and my stereo system.

because the telephone provides them (this circuit is designed so that it disturbs the operation of telephone as little as possible so it has high impedance input).

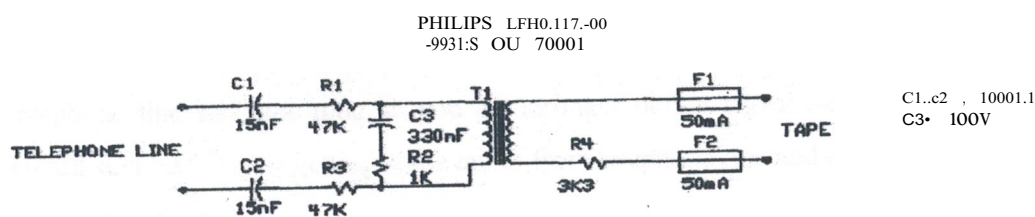


Figure 3.3

I found out the type of all other components than the transformer T1. T1 is the audio isolation transformer, which seems to have properties quite similar to typical 600:600 ohm telecom isolation transformer. The components F1 and F2 are 50mA fuses. The signal output level is suitable for microphone input because the resistors attenuate the voice signals from telephone line around 40 dB (some telephone equipment regulations in Finland needed this).. On the picture below you can see a picture of the inside of the Philips LFH0117/00 telephone recording adapter: Show in figure 3.4



Figure 3.4

In the circuit the two 15nF capacitors are blocking the line DC level and the low frequency for ringing. All other components are safety requirements, for lowering the noise and for matching the telephone equipment regulations (signal isolation from line, impedances etc.). This schematic is basically a very safe one: fuses are not necessary for proper function (just for extra safety) and the transformer provides galvanic isolation from telephone line; The circuit is designed to be connected to the microphone input of a recorder. (the output signal level is typically few millivolts: which is too low for any other type of input).

3.6 Noreico Recording Interface

The second circuit is made by Noreico and is also designed to be used in parallel with existing telephone. This circuit has much less attenuation between telephone line and tape plug so you get stronger output signal. I have used this circuit successfully for getting some audio from telephone line and sending some audio back to telephone line:

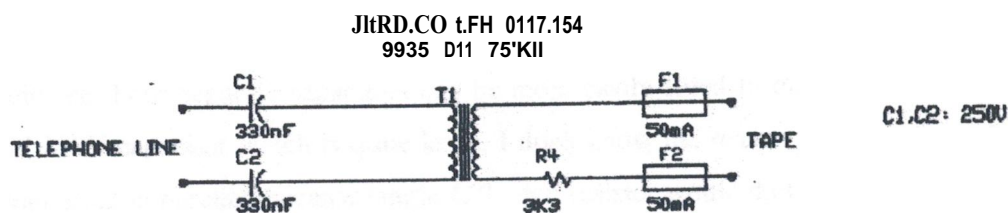


Figure 3.5

I found out the type of all other components than the transformer T1. T1 is the audio isolation transformer, which seems to have properties quite similar to typical 600:600 ohm telecommunications isolation transformer. The components F1 and F2 are 50mA fuses.

3.7 Recording Interface From Tekniikan Maaailma Magazine

The third circuit was shown in an article written by Martti Koskinen in Tekniikan Maaailma magazine issue 8/1994 pages 94-95. The circuit is designed for recording telephone conversations using normal tape recorder. The circuit has an option to also play back sound to the telephone line from separate connector. The transformer T1 is typical 600:600 ohm telephone isolation transformer with centre tapped secondary. This circuit is also designed to be used in parallel with existing telephone.

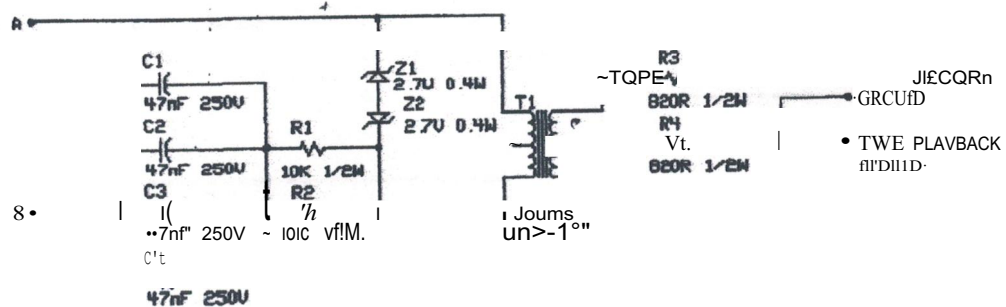


Figure 3.6

The capacitors C1 to C4 are connected in parallel to make about 200 nF capacitance. Four separate capacitors can be more easily fitted to one case than single 200nF 250V capacitor which is quite large. I don't know the reason of why R1 and R2 are connected in parallel, because single 4.7k ohm resistor would do their job as well.

3.8 Marantz PMD Recorder Series

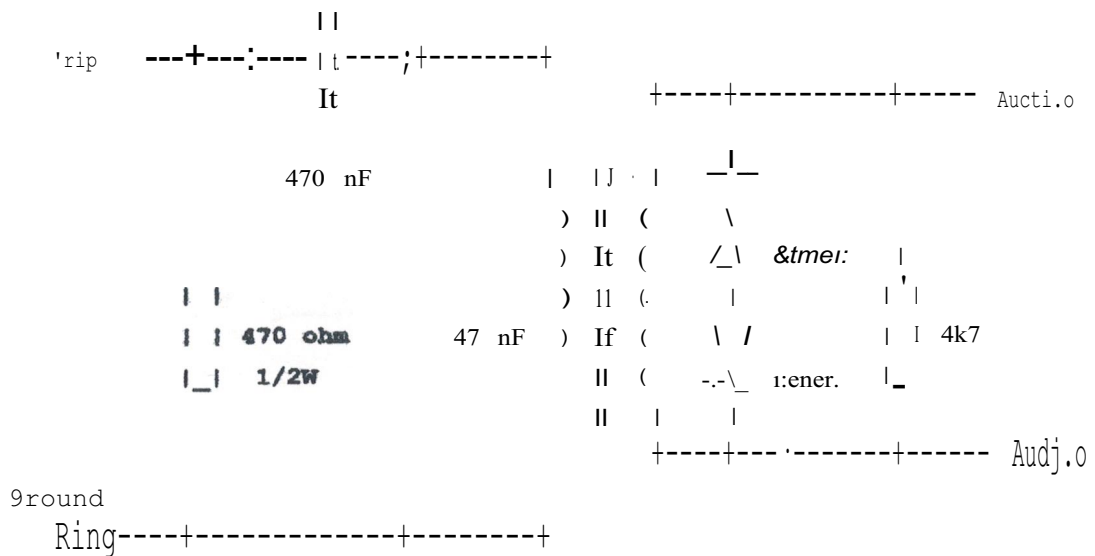


Figure 3.7

Tip and ring are first shunted with a 470 ohm 1/2 W resistor (to allow the interface to sieze the line). Next, before the transformer primary, there is a 470nF series cap (high pass, and DC transformer isolation) and a 47nF shunt cap (low pass to limit the upper end). The transformer is a 1:1 600 ohm, I believe, though the schematic doesn't specify.

On the secondary there is first a shunt pair of back to back Zeners to limit the max voltage seen by the recorder circuit, and then a shunt 4.7K ohm to ground.

3.9 Telephone to Studio Mixer Interface

The transformer and 44 nF capacitor keeps the impedance seen from line high enough that not bad mismatching happens when connected to studio mixer. If I would be connecting something like this to my audio gear I would add some type of surge protection to the circuit (two zener diodes in output would be nice) or add external surge protector. But let's the original text to describe the circuit in more detail.

We use telephone audio in our studio all the time. And yes, it's an off the shelf design. I designed and built such a device with scrap door components. I used an audio

coupling transformer and a capacitor. The primary windings add in series to 500 ohms. Instead of connecting them directly together I added a cap between them. It was something like 0.047 micro farads with a 600 volt rating. And the secondary which is 500 ohms runs into the control room mixer.

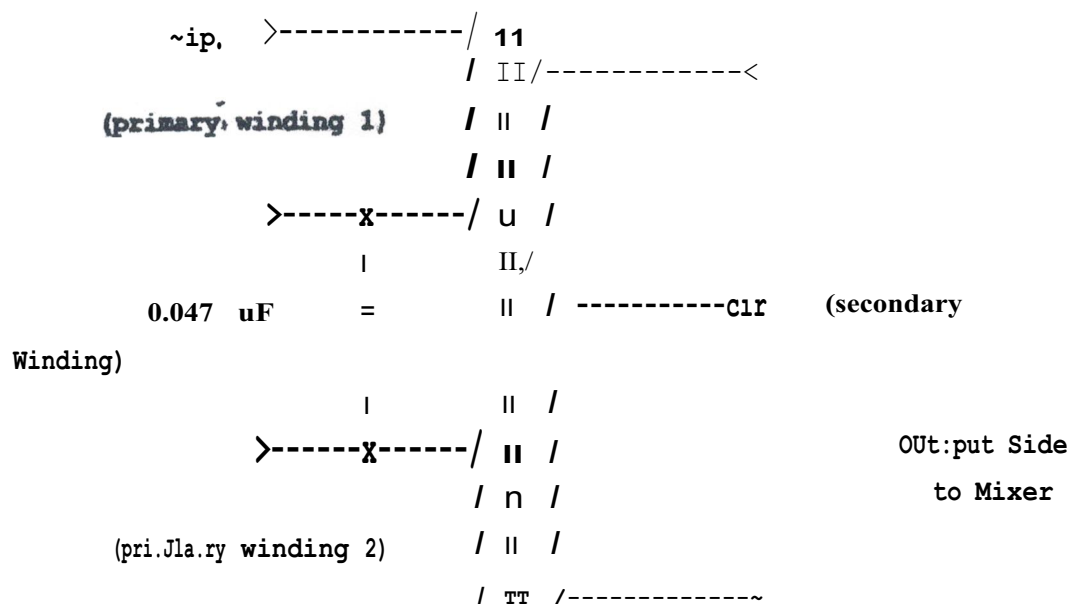


Figure 3.8

This circuit works great for us in the studio. The circuit is designed to be used in parallel with existing telephone. Just make sure you use properly rated components.

3. Audio interfaces without Transformer Isolation

In some special cases the audio interface is built without isolation transformers. In these cases the audio signal is passed from telephone line through the capacitor which blocks the DC from telephone line. This type of isolation works quite well in applications where you don't want to use the transformer but you still want to get some audio from the line. Typical application is called 'ID boxes'.

A typical capacitor isolation circuit:

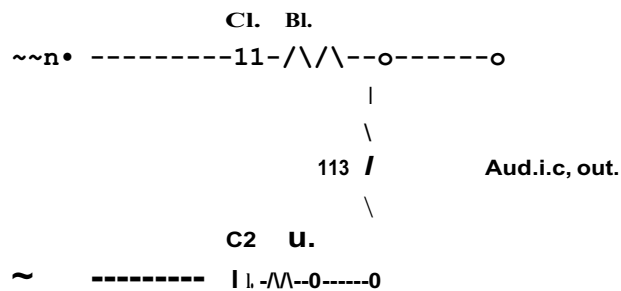


Figure 3.9

The capacitors C1 and C2 block the DC and pass the audio signal to the output. The resistors R1 and R2 provide some protection against the spikes on the telephone line, and the circuit is designed to handle the 1.5 kV pulses. It does not disturb the telephone line operation. R1, R2 and R3 make together a voltage division network which will attenuate the audio signal coming from telephone line to the desired signal output level.

The circuit should be connected to differential audio input. If the circuit is connected to single-ended input the circuit works worse and gets easily all kinds of interference. Capacitors C1 and C2 should be rated to handle the 1.5 kV pulses. The capacitors C1, C2, R1 and R2 should provide a high impedance to the telephone line so that the telephone line balancing is not disturbed. You should also note that this circuit does not provide as good surge protection as a transformer (surges can quite easily pass through C1, C2, R1 and R2). This is not the preferred way to do the telephone line interface! Preferred way is to use transformer isolation instead.

3.11 Simple Teleom Hybrid Circuits

Telephone hybrid circuit is the circuit which is designed for converting 2-wire interface to 4-wire interface and is one of the basic building blocks of the telephone system. Telephone hybrid is the circuit which separates the transmitted and received audio which are sent both at the same wire pair in 2-wire normal telephone interface.

There are many different types of hybrid circuit in use. Traditionally telephones have used combination of special transformer and few additional components to keep incoming and outgoing signal separated from each other. Nowadays this is done more or less electronically.

In telephone central end hybrid circuits are needed when must be done any

amplification to the signal. Traditionally, the systems separate the incoming and outgoing signal, then they are amplified separately and sent to other telephone central using separate wires or otherwise separate communication channels. The oldest models of those circuits have been built from one or two transformers and some other balancing components to get best results. The problem has been how to get good balance to the hybrid circuit, said in other way how to separate incoming and outgoing signals as well as possible. Nowadays everybody is avoiding bulky and expensive special transformers, and more and more electronics are used because it is cheaper. Modern hybrid circuits consist only of one audio isolation transformer, two operational amplifiers, resistors and some capacitors, and the most modern approaches try to avoid that transformer altogether by using active electronic circuits. In telephone line side to do the job, and opto-couplers to do the isolation where needed.

Many different system circuits have been used and I am showing here just one basic transformer-based circuit which is easy to understand and is useful for many experiments.

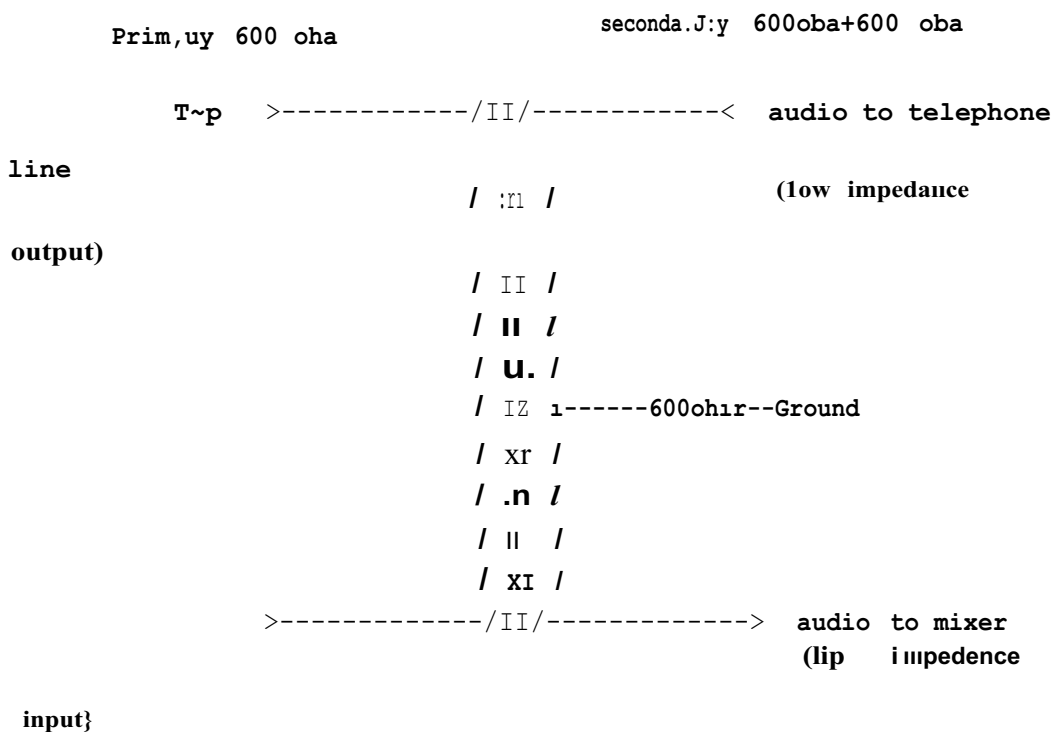


Figure3.10

This first circuit is a traditional simple hybrid circuit which have been earlier

successfully used in many telephone circuit (for example modems). The circuit works so that the 600 ohm resistor in the center pin of the secondary is seen as 600 ohm impedance load in primary circuit. The end of the secondary which is connected to low impedance audio output (for example amplifier made-for-driving 8 ohm speaker; must be always connected to amplifier or ground to make the circuit work as expected. The audio signal output from the circuit must be fed to high impedance (>10 k ohm) audio input to make sure that the operation of the circuit is not disturbed. The circuit gives quite acceptable separation between incoming and outgoing signals when all impedances are set correctly. The 600 ohm impedance is kind of idealistic value and does not fully reflect the reality. In real life the impedance of the telephone line or telephone is not exactly 600 ohm and the transformer has its losses. A 600 ohm resistor is anyway quite a good starting point.

If transmitted and received signals mix with each other, you will have to fiddle with the balancing network. For experiments I can suggest fitting 1 k ohm variable resistor to the place of 600 ohm resistor for experimenting which impedance value gives the best results. You may also want to try other type of line impedance simulation circuits if you know what matches your system better. If the impedances presented by both the send and receive sides are the same the hybrid circuit will work quite well. You will find that the send and receive signals don't interfere with each other, but both come and go from and to the line.

If you are thinking of connecting this circuit to telephone line or otherwise sending OC current through the primary of the transformer remember to use a transformer which can handle the DC without saturating (telephone transformers made for "wet" circuits). And remember that there are strict rules what the equipment you connect to telephone line must meet and you are not allowed to connect anything not approved to public telephone system.

3.12 Modified Circuit

The following circuit is for telephone line interfacing when using a 600 ohm to 600 ohm transformer with center tapped output:

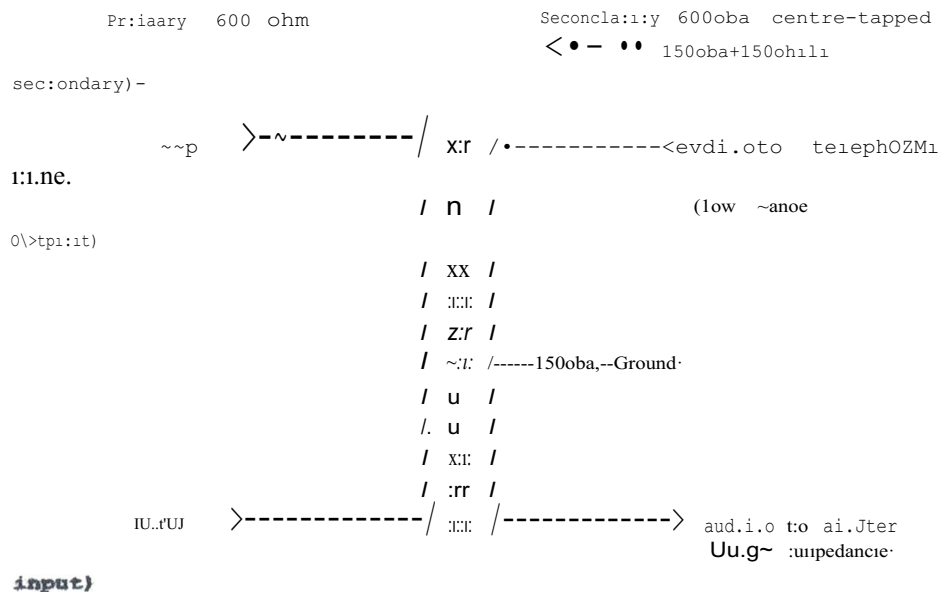


Figure J.11

This circuit works the same as that circuit above, but uses standard telephone line transformers easily available. One common transformer type has 600 ohm primary and 600 ohm centre-tapped secondary. In centre-tapped secondary each secondary side presents 150 ohm impedance. By using 150 ohm resistor connected to secondary centre-pin, the primary sees 600 ohm impedance. Experimenters can try for example 470 ohm variable resistor instead of 150 ohm fixed resistor to test which value gives the best results. You can see this type of circuit built to a small plastic box at the picture below:

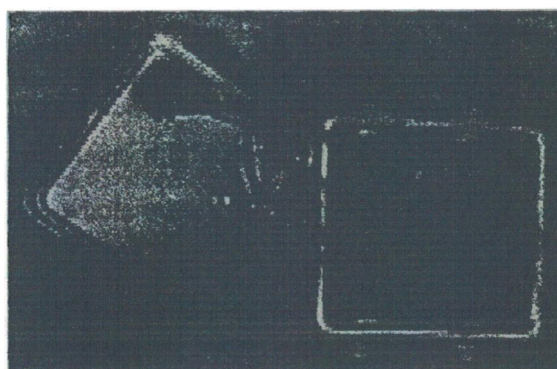


Figure J.12

3.13 Soundcard To Telephone Line Interface

The following circuit is a modified version of the circuit above. This circuit is designed to provide audio output which can be fed to a PC soundcard and an on/off hook switch:

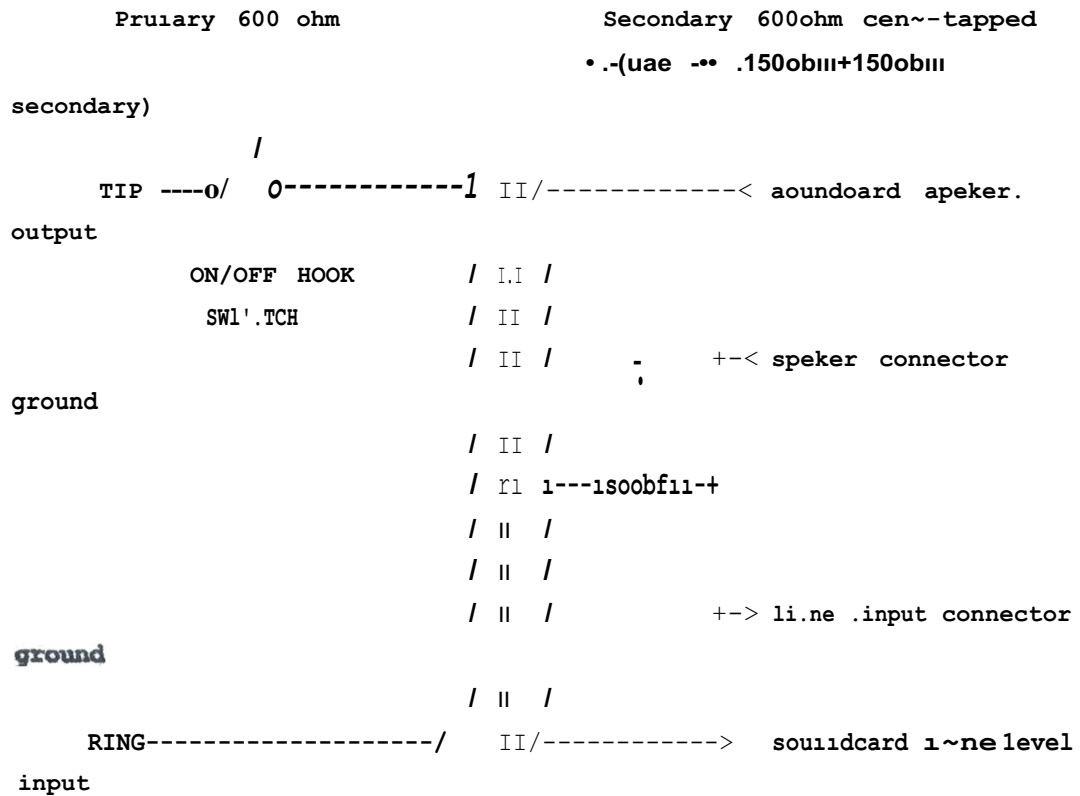


Figure 3.13

Remember that in telephony applications, the signal levels must be adjusted carefully and must be limited. The circuit is, in good enough, balanced that there is no annoying feedback in the whole system. Warning that this circuit is a little bit simplified interface diagram. This diagram is for example overvoltage protection on the audio output and also limiting circuits which stop too large signal levels to enter the telephone network.

3.14 Operational Amplifier Based Hybrid Circuits

Modern modems use hybrid circuits built from operational amplifiers-resistors and a 600 ohm isolation transformer. With a 1 amp transformer the hybrid circuit can be made cheaper and, perhaps, more accurate.

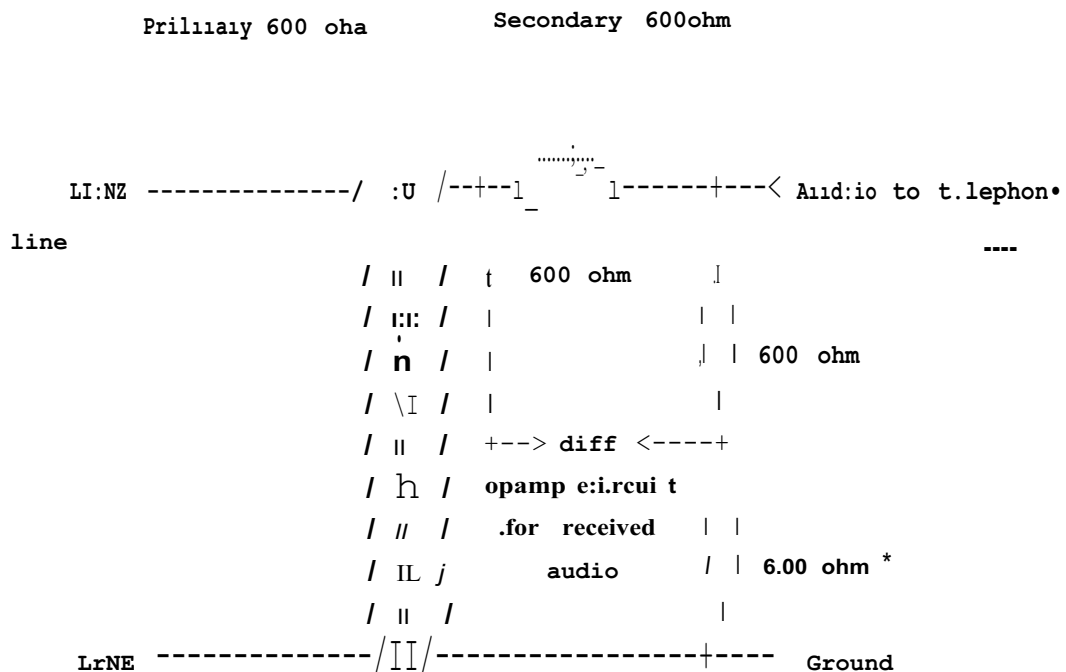


Figure 3.14

The difference for audio signal which is transmitted to the telephone line should be low impedance to ensure that the impedance matching to telephone line is correct. For receiving audio a differential amplifier must be used to separate the incoming signal from outgoing signal but differential amplifier is very easy to implement using operational amplifier. The performance of the circuit can be made better by replacing the 600 ohm resistor which is marked by with some better model for the telephone line seen through the isolation amplifier. A better model provides better isolation between incoming and outgoing signals.

A quick route to mixing desk users: professional mixing desks nowadays have 4 inputs and low impedance inputs. This makes it very easy to connect with this type of circuit if you happen to own a good audio mixer.

Here is the full operational amplifier based hybrid circuit diagram (theoretical circuit):

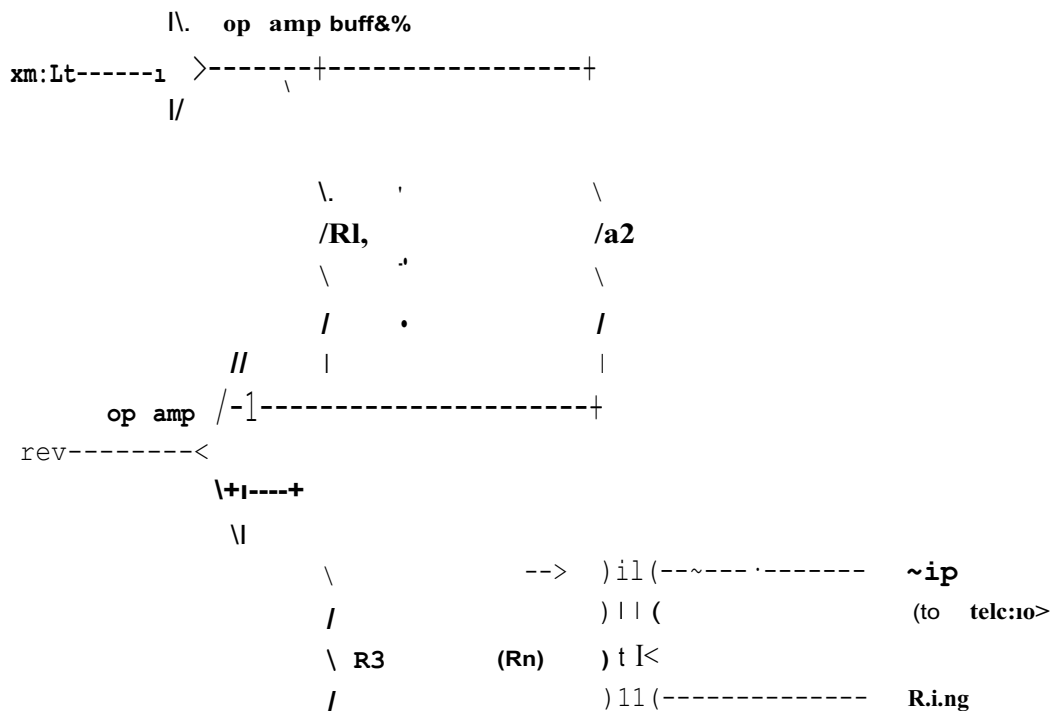


Figure 3.15

In-the above diagram R_n = the telco network impedance as seen at the other side of the transformer.

This circuit is an example of an "active" hybrid. Essentially it is a balanced network. If the ratio of $R_1/R_3 = R_2/R_n$, then you have infinite return loss - that is, you should have none of your transmit signal appearing on your receive line (when this happens, this signal is called side-tone). Yet the receive signal from the "far-end" will appear on the receive line. In other words, two signals can use the same two-wire interface, yet are separable. Note that the resistors which define the amplification of the op-amps are not drawn here. so if you are planning to build this circuit you will have to add them.

Unfortunately, Telco line impedances can vary quite a bit, so the ratio of R_2/R_n rarely equals R_1/R_3 except in situations where the designer has tight control over loop lengths and terminations. Any imbalance in the balanced network creates side tone - a small amount of the transmit signal will appear on the receive line. In typical situations

the side tone can be attenuated around 20-30 dB with a well designed hybrid circuit.

Another typical way to implement a hybrid circuit is to build an opto amplifier

Circuit which takes the signal over the transformer coil and subtracts the transmitted signal from it. The following circuit diagram shows this:

NDIE
TMNSIWTTT

$$\{I_t \cdot R_2\} \quad R_t \quad 600$$

$$I_t$$

TD MODEM:
RECEIVER

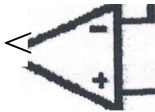


Figure 3.16

The circuit below is, partly redrawn optimized, hybrid circuit from National Semiconductor application note "Optimum Hybrid Design" from 1-98'5 (that application note is no longer available),

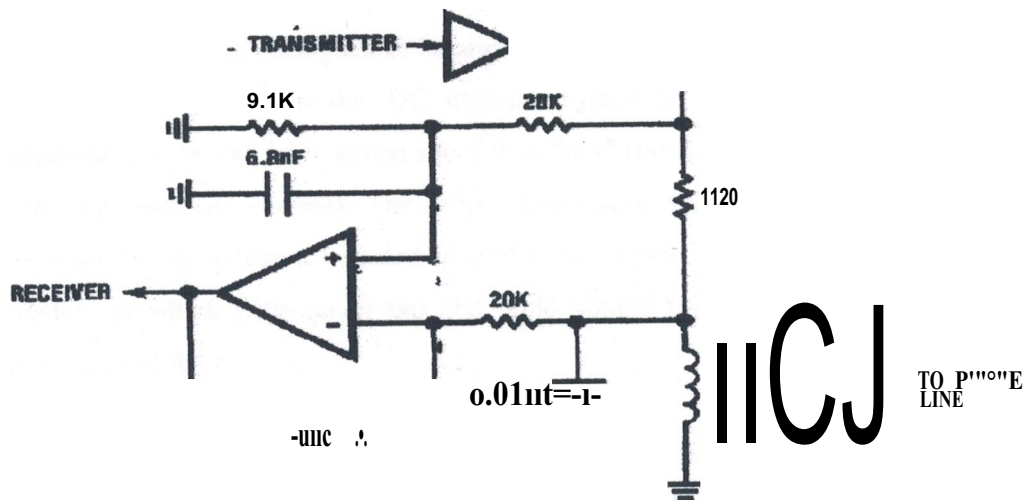


Figure 3.17

The transformer in this circuit is 600:600 ohm telephone line transformer. For best results you have to adapt the component values slightly to match the line impedance and the transformer you are using:

That upper amplifier (the triangle with one input and output wire) is just a buffer amplifier with an amplification factor of one. Signal from transmitter is connected to the positive input of opamp. The negative input of that op-amp is connected to the op-amp output.

3.15 More details: Impedance Matching Telephone Line Interface

Telephone line interface has to provide two functions when it is off-hook:

- Provide DC path for current flowing in telephone line. Normally, there flows about 20-50mA current in telephone line and telephone regulations typically specify that the DC resistance must be less than 400 ohms.
- Provide proper termination for telephone audio frequencies (300-3400 Hz). This is typically specified to be 600 ohms.

3.15.1 Wet transformer

Traditionally those two functions are accomplished in modems and telephones by "wet" telephone transformer. Wet type means that the transformer is designed to handle the DC current (typically 20-50mA) properly and does not saturate at this DC current. Typically "wet" transformers are more expensive, bigger and have worse specs than "dry" transformers (which do not have to withstand any DC current). The proper termination in modems is provided by the electro-mechanical circuit connected to the transformer. Another possibility is to use a transformer which has a center tap and build a simple transformer and resistor hybrid circuit around it.

Here is a typical circuit for "wet" transformer:

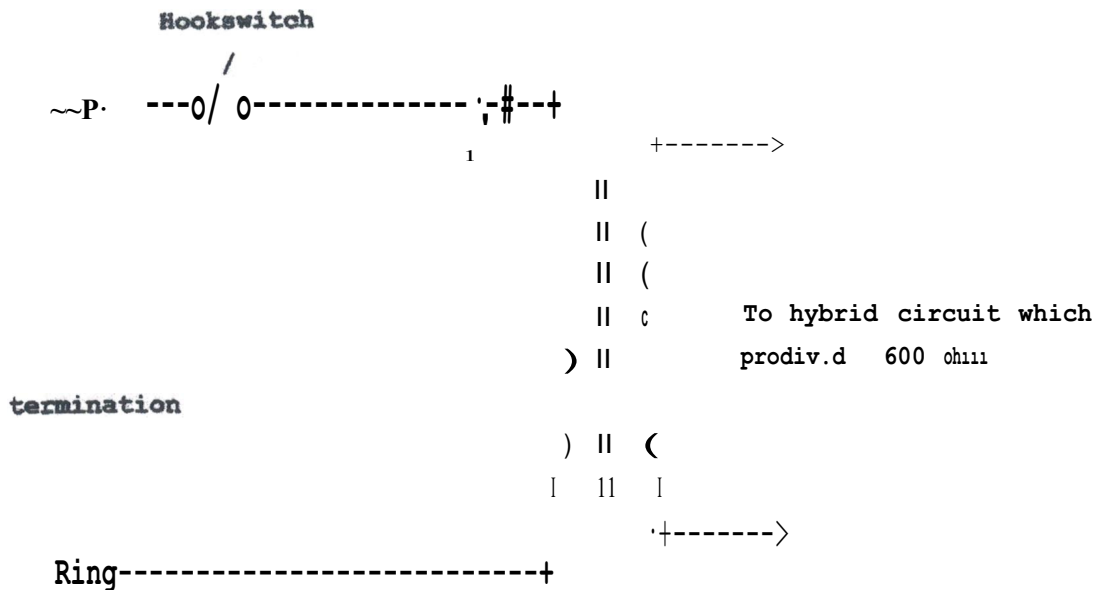
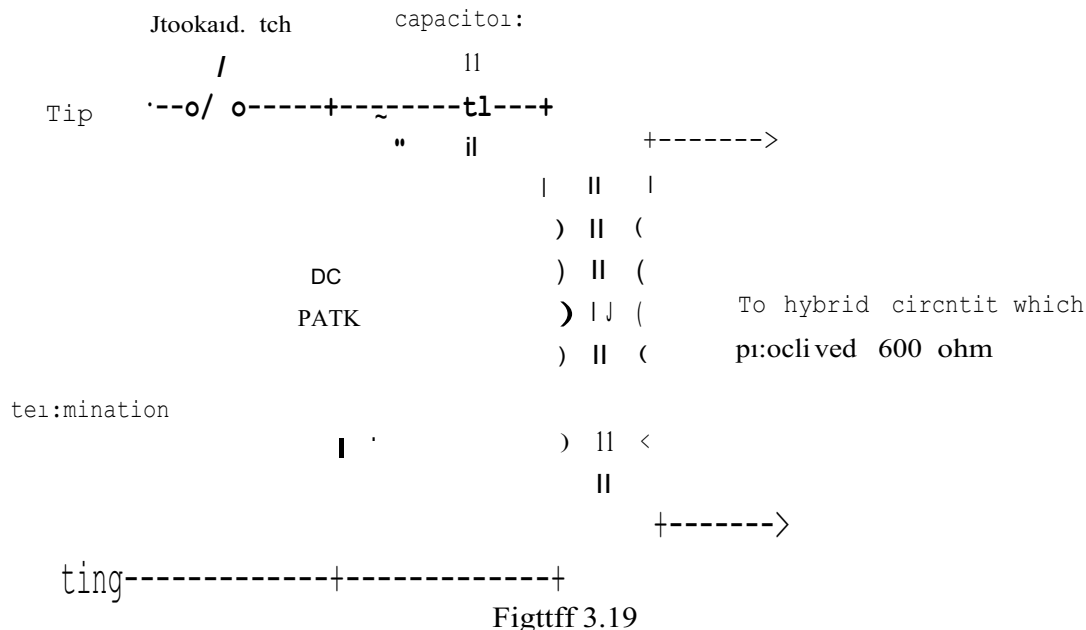


Figure 3.18

The circuit operation is quite straightforward, when the hookswitch is closed the telephone-line DC current starts to flow through the transformer primary coil. The DC resistance of the circuit is determined by the resistance of the transformer primary coil (typically in 60-200 ohm range in 600-600 ohm telecommunication transformers). The transformer is 1:1 transformer designed to operate at 600 ohm impedances, so 600 ohm termination provided in the secondary reflects as 600 ohm to primary (this is not totally accurate because transformer has some losses so you need a little smaller than 600 ohm resistance in secondary so that it looks 600 ohms in primary).

3.15.2 "Dry" transformers

Dry transformers are transformers which are not designed to handle DC current flowing through them (if you put DC through them they saturate and do not work correctly as transformers). 600:600 ohm dry transformers are very useful for example in modems because they are available in small sizes (even so small that can be fitted inside PCMCIA modem card) and can have very good performance figures. Because the dry transformer can't stand DC then in telephone application where there is DC present the DC must be blocked by suitable capacitor (usually 2.10 uF) and alternate path for DC must be provided. Here is an example circuit:



The DC path must be designed so that it will pass DC well but provides high impedance to telephone audio frequencies (so that it does not disturb the impedance matching done elsewhere). A large inductance coil can be used in this but it is not practical because you wanted to get rid of that hulky "wet" transformer using small "dry" transformer instead, so you don't want an expensive and bulky coil in your circuit.

Fortunately coils can be simulated electronically using gyrator circuit. With it is very easy to have a simulated coil which has low DC resistance and the circuit looks like high inductance: cell (few H.m.i.e.s. simulated: coil can be made easily). Another possibility is to use constant current sinking circuitry. Constant current circuit provides path to DC current but has very high impedance (before using constant current circuitry take a look if your telephone regulations allow constant current operation or you can make the circuit to work inside the specs in varying line conditions).

When you add electronics to transformer primary side remember that those must work at both line polarities. A bridge rectifier will help to make sure that the current going to DC path circuit is always at correct polarity. Another thing to consider is overvoltage protection because your circuit in the transformer primary side has to withstand the spikes which exist in telephone lines primary side. Make also the circuit

so that it is not damaged by a little more current than normally present in telephone line (sometimes the reverse is true) and you don't want to break down too easily).

3.16 Problems in Linking Telephone Hybrid to Audio System

Even the best engineer has had the dear wife telephone: In essence, at one time or another, linking the phone conversation to audio system. Here taking a Us-to-audio studio can be more problematic than you think; still, though, you can get a good view of the scenario at article *Phone Line Has Problems*, article from *JK Audio*. The final problem in making the audio connection is that the of the "line" pin (transformer, a full hybrid, digital hybrid, etc.), to the caller, and back to the studio into the telephone line input of the phone patch. (You can hear this leakage in the earpiece of your telephone handset. Just , listen at 4:10 @ www.ucl.ac.uk/fy.omic.GW11 , voice ~ m. G. Shalek to you!)

3.16.1 Level Matching on Local and Remote Voice

Typical commercial telephone hybrid allows the equalizing of telephone line is full-duplex interface implemented using single twisted pair. When an announcer speaks, his voice travels through the phone line output feeds of local and remote voices. Typically a hybrid needs adjustment for every new connection because of impedance changes. Today automatic digital hybrids are available for equalizing local and remote telephone conversations.

3.16.2 Trans-hybrid loss and announcer voice distortion

Trans-hybrid loss is that portion of the announcer's voice that leaks through the hybrid to its audio output. The higher this speed, in dB, the better isolation in the device. This leakage is distorted and phase shifted after its long journey. In the studio, the announcer and is mixed at the console with the phone patch (caller), to create the on-air mix. When you use a poor phone patch, its output includes a distorted, phase shifted version of the announcer signal. When this leakage is combined with the clean announcer material, the "unhybrid" sound is as follows: *as follows, it is not affected by phase cancellation than others: The greater the trans-hybrid loss, the less announcer audio that leaks into the hybrid output and the less the announcer voice*

,distortion. Ideally the output of the hybrid should consist of caller and only a Digital
11;4r-M-s have suggested that T-oe-s-sing electronics to 'get better kafiHlybr-icl less -figures lhatti
which are available with simple analogue solutions; you have to decide what's best for
your budget. There are different requirements depending on the application (broadcast,
teleconference or remote training).

It has been suggested that ISDN be used as its full duplex is good one, but it might
be only practical if both sides of the two-path have ISDN: When calling, between
plain old telephone services (POTS) and, ISDN, the above problems remain.

3.1-6.3 Echo Problem in Long Distance Calls

Echo is caused because of the coupling between incoming and outgoing audio
in the telephone circuit and the delay in the telephone line (especially in long distance calls);
Echoed back audio is usually caused by an impedance mismatch at a 214-wire conversion point
(such as a codec-annex-hybrid, analog CO line interface) and by acoustic feedback. Thus there
is echo: ISDN or other digital telephone set on an all-digital connection: would

Not cause echo because of conversion mismatch; but if normal handset or hand,
free telephone is used the acoustic echo is still possible.

Echo doesn't become audible until the delay in the circuit exceeds a certain
threshold value which depends on the losses in the circuit. Even milliseconds of
terrestrial echo can be annoying, but typically the delays are not annoying if the delay stays
below 25ms. Old Bell standards said that on calls of more than 1800 miles, an echo
suppressor was used. In general, you need echo cancellation when the delay exceeds
some subjective value in the 30-50 ms range.

As it is practically impossible to prevent echo (by perfectly matching the
impedance in line circuits and by acoustically insulating all phones), it either has to be
suppressed or cancelled when it does occur. For this reason, echo cancellers are
deployed by Telephone Company on long-haul routes that; when used, bring the total
circuit delay to above the echo threshold value determined by line loss. These echo
cancellers are deployed on both sides of such long-haul routes and the echo canceller at
the remote end of the call is responsible for ensuring that you don't hear any echo.

For more information on how echo canceling works, please consult ITU-T
Recommendation G.165 or some good telecommunication book. The morale is therefore
that if you hear echo, you can't do practically anything about it, as both the cause of the

problem and the solution to it: lie at the remote end of the connection (typically at the telephone company's equipment). In the connection you're talking about is across a private network make sure that the echo cancellers are correctly dimensioned because wrongly dimensioned echo cancellers will be totally ineffective.

3.16.4 Metallic-Sounding Caller Voice Problem

If your telephone connection is through a digital PBX or digital switch (typical in many areas) there might encounter a problem with the voice which might sound OK on telephone but sound "metallic" when you connect it to the mixing desk through your high-quality hybrid circuit. The metallic sound problem is an aliasing problem caused by the digital telephone system where there is not much filtering after the DAC converter which produces the sound. The absence of the output filters causes that there are high frequency noise components added to the output audio signal. The audio sounds fine on normal telephone because it can only play back the normal telephone audio. The problem is audible with a hybrid circuit of a hybrid circuit has wider bandwidth than a normal telephone. The solution to make this signal sound normal telephone is to remove everything above 4 kHz by a sharp low-pass filter. You can try if your mixing desk has a parametric equalizer, and if it does enough, to roll off the high frequencies, this problem. When you equalize the signal from telephone hybrid then you can also remove the bass frequencies also: (there is usable sound information below 200 Hz on normal telephone line) so you can also get rid of the possible low frequency noise (mainly 50 Hz or 60 Hz) which is common in the telephone line.

3.17 Helpful Tips for Telephone Hybrid Circuit Designers

3.17.1 Measuring Return Loss figures

Regardless of how you measure the mismatch between the impedance of the line and the termination or load is Z_L then the return loss is given by the formula:

$$RL = 20 \cdot \log \left\{ \frac{|Z_L - Z_0|}{|Z_L + Z_0|} \right\}$$

The log function in the formula above is the logarithm of RL.

The return loss must meet the regulations in the whole specified frequency range. The measurements can be quite easily made using a variable frequency signal sine wave generator and the reference impedance Z_0 (can be built easily from resistors and capacitors). The following circuit can be used to measure the return loss:

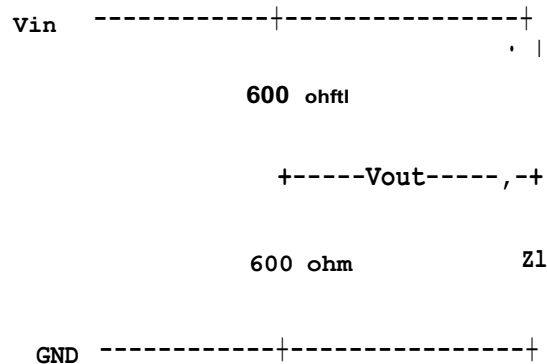


Figure 3.29

If you want to test the device with the signal level of V_m then you put the voltage $2 \cdot V_{in}$ to the circuit from the signal generator (the input impedance of the circuit is

Around 600 ohms if Z_0 and Z_t are near 600 ohms). Connect the reference impedance Z_0 and the measured telephone interface circuit Z_l to this measurement circuit. Connect millimeter to the circuit to place marked with V_{out} to measure the V_{out} voltage. In ideally balanced circuit this voltage is always zero. Make sure that your millimeter can measure the AC voltages in the frequency range you are using accurately (some millimeters have very large measurement error when frequencies go much higher than few hundred Hz).

Using the circuit is very simple. Just apply the input signal and measure the output voltage. Do as many measurements as necessary to cover the whole specified frequency range. When you have made the measurements you can calculate the return loss using following formula:

$$RL = 20 \cdot \log_{10} \left(\frac{V_m}{V_{out}} \right)$$

If you want to measure telephone equipment which need some DC current flowing through the circuit you try to measure you have to use a little bit more complicated circuit to be able to separate the DC signal from the measurement circuit using capacitor (the capacitor does not cause much error on telephone 300-

3400 Hz -frequency range, for -lower frequencies use higher value). The power to the measured telephone or other equipment must be fed from separate power supply And run through AC block circuit which prevents the power source for short circuiting the AC signals-. This AC blocking circuit can be a large coil [preferably more than 5 henries]- gyrator circuit or constant current source.

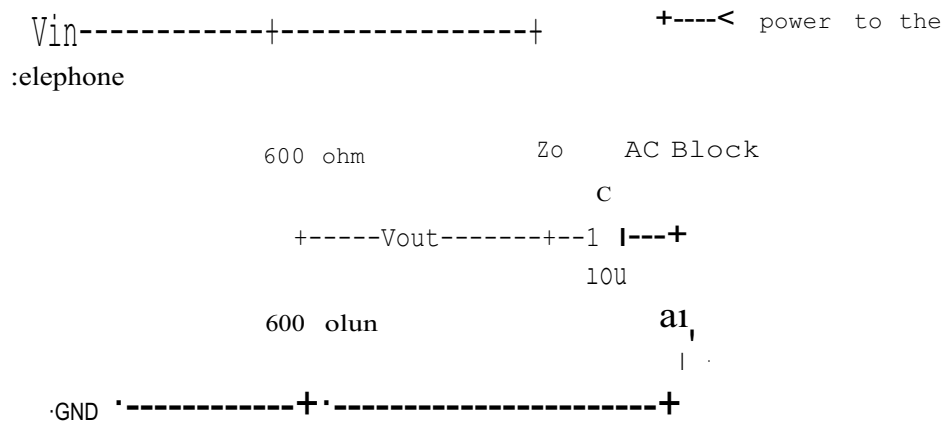


Figure 3.21

The measurements can be done with this circuit in the same way as the original circuit. The only thing you must consider is the possible measurement errors caused by the capacitor and AC blocking circuit. To make sure that $Z > 1/\omega \pi \cdot f \cdot C$, $H\}$ uF is a good value to start because it has maximum resistance of about 50 ohms in the telephone audio spectrum (100-3400-Hz).

3.18 Simulating telephone line

3.18.1 Resistor and capacitor network simulation models

The most traditional way to simulate telephone line is to use resistor and capacitor networks to simulate the attenuation caused by the telephone line. A typical model for this type of telephone line simulator is a resistor and capacitor network which looks like This:

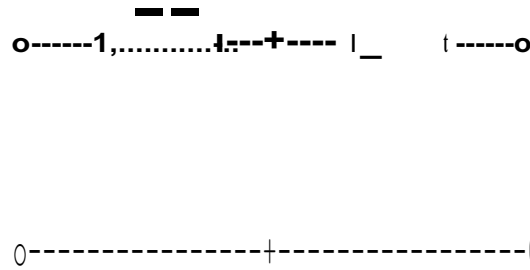


Figure 3.22

The resistor R and capacitor C values depend on the cable characteristics. Old Swedish telephone equipment regulations have listed the following values for simulating a typical

40cm loop cable:

Length	Cable diameter		
0.5 km	0.4 mm	70ohm	10' nF
1.0 km	0.4mm	140 ohm	40 nF
1.5 km	0.5 mm.	210 ohm	20 nF
2.0 km	0.5mm	280ohm	40 nF

The circuit can be modified for simulating symmetrical cable better in some measurement by dividing the resistance to four wires. This arrangement leads to the following circuit:

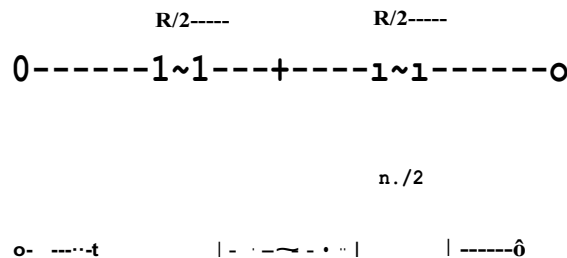


Figure 3.23

3.18.2 Build a Telephone Line Test system From Telephone Cable

Find an old spool of 25 pair cable, preferably pulp insulated, from the back of the warehouse. Punch down each end to opposite sides of a 66 block, only on one side start with pair *two*, and bring pair one down to the last two terminals, after pair 25

Stick in bridging clips across all but the bottom pair of the block. In this way you get quite easily very long line to test, quite easily. Attach one end of this to a cheap phone line simulator (all it needs to provide is battery, dial tone, and ring voltage.) Buy a couple of test dips from your usual supplier, and now you have a fairly easy to use test device for cheap,

If you need to simulate the interference which can go to the cable, use an old office fan and an interference source by putting this right next to the spool of cable and turn it on during testing.

If you are doing worst case testing, you should use junky cable from the recycling bin and stick in loading coils at the appropriate intervals. To be really nasty, take a couple dozen feet of your telephone this cable, create leaks in the cable and put that cable into water (you can add some salt and dirt to the water for more realistic situation). Add this into the middle of your test circuit someplace. Now you have something that is beginning to approach the real world cable plant worst case.

3.19- Testing Standards,

To be able to do any repeatable testing one must have control over the equipment and the test environment. Aim to create an environment that is not influenced by other individuals, one must test to a given set of "standards". For USA standards TSB17 A and TSB38 deal in telephone lines and modem testing (Loop 1 condition for them is EAI 1, which is a 2kft of 26ga). For longer loop there can be up to 5 loading coils in the line. Other countries have also standards on the testing conditions but I have not found references to them.

3.19.1 Other Technical Regulations For Telephone Line Terminal.

European standards

The following specs are taken from the European NE-T4 (ETS 300 001, second edition- April 1994) regulations. ETS 300 001 is, basically, a big collection of the various European countries parameters I have put some parts of the specs (Finnish part) to here:

On-hook

- " DC resistance must be at least 1 M ohm when measured at 100 V voltage
- „ Isolation resistance must be at least 5 M ohm from line to touchable metal parts measured at 100 V voltage
- The impedance in voice frequency (200-3400 Hz) must be greater than 10 k ohm when measured with 5V RMS audio signal

Off-hook

- Isolation resistance must be at least 5 M ohm from line to touchable metal parts measured at 100 V voltage
- DC-resistance must be less than 400 ohm in current values between 20 mA and 50 mA
- If the terminal equipment used: constant current principle then the current must be in 20-50 mA region under all conditions
- The impedance of the terminal equipment must be so matched that the return loss is greater than 10 dB compared to 1000 ohm reference
- When the terminal equipment transmit voice or music the mean signal level must not be greater than -10 dBm level in any 10s timeslot
- When other signals are sent to line the signal must not be greater than -10 dBm in any 200 ms timeslot
- The average level between 1400 Hz and 12 kHz must be attenuated 12 dB/oct and the signal level in frequencies greater than 12 kHz must be less than -55 dBm
- Common mode rejection must be greater than 40 dB in 40-300 Hz region, greater than 50 dB in 300-600 Hz region and greater than 55 dB in 600-3400 Hz region.

Note on signal levels: 0 dBm means 0.775 Vrms level, so -10 dBm is around 0.2 Vrms.

3.19.2 Equipment in Series with Telephone

- " The series resistance must be less than 200 ohm:
- The attenuation of audio signal must be less than 1 dB at 800 Hz

There are also many other technical specs in NET4 document, but those are the most critical to hybrid 'Circuits when building telephone signals you should also consider the telephone equipment: electrical safety regulation in EN 41 003 standard. This means generally that the equipment must withstand 2.1 kV surge and DC between the telephone line. The equipment does not be able to cause dangerous voltages to the telephone line or to touchable parts in any probable single component failure. And many other safety- regulations.

4. Sources of Failure in the PSTN

What makes a distributed system "reliable"? A study of failures in the US Public Switched Telephone Network shows that human intervention is one key to this large system's reliability. To operate successfully, most large distributed systems depend on hardware, communication, and software—and any one of these elements can disrupt or bring down an entire system. One such distributed system, the US Public Switched Telephone Network (PSTN), is the US portion of possibly the largest distributed system in existence. [1] Like all telecommunication, switching, and the PSTN present a fairly complex task. It connects point A with point B. Paradoxically, this seemingly trivial task requires some of the most complex and sophisticated computing systems in existence. Software for a switch with even a relatively small set of features may comprise several million lines of code.

The PSTN contains thousands of switches. Switches include redundant hardware and extensive self-checking and recovery software. For several decades, AT&T has expected its switches to experience no more than two hours of failure in 40 years [2] a failure rate of 5.7×10^{-4} . Since 1992, telephone companies have been required to notify the US Federal Communications Commission (FCC) of outages affecting more than 30,000 customers. I used these outage records to determine the principal causes of PSTN failures. To account for the possible effects of seasonal fluctuations in call processing volume, I analyzed failures over two years, from April 1992 to March 1994, beginning with the earliest FCC reports. I made a questionnaire to determine how each outage affects system dependability, in an effort to shed some light on the dependability of different components (including software):

Major sources of failure were human error (on the part of both telephone company personnel and others), access infrastructure, and overload. Jvarkian caused nearly half of all downtime (44 percent) in ten seconds of outage minutes,

An unexpected fitidfog, given the complexity of the PSTN and the heavy reliance on software, was that software errors caused less system downtime (2 percent) than all other sources of failure except vandalism. Hardware failures were similar to software failures in terms of average number of customers affected (96,000 and 111,066) and duration of outage (160 and 119 minutes).

Errors on the part of telephone company personnel and acts of vandalism caused similar amounts of downtime (1.8 and 1.8 percent).

4.1 Failure Classifications

Table 1 shows the failure classification scheme I used, a scheme that is general enough for comparisons with failures in other large distributed systems. In the case of the human error category, I separated errors made by telephone company personnel from those made by other employees because the companies have direct control over employees only. Overload conditions are accounted for separately because they represent failures accepted as an engineering trade-off between dependability and cost.

Table 4.1 Failure categories

Category	Source	Examples
Human error	Errors made by	Errors in
Company	telephone company personnel	<ul style="list-style-type: none"> * cable maintenance * power supply maintenance * power monitoring * Facility or hardware board maintenance * Software version mismatches * Following software maintenance procedures {Such as errors in patch installations and Configuration changes; does not include source code changes} * Data entry
Human error	Errors made by people	Cable cuttings

Others	Other than telephone	Accidents (for example, cars striking telephone poles, etc.)
Major and minor cable	Major and minor cable	Cable, power supply, or facility damaged from events, but wiring, terminals, or lighting
Natural disasters		Earthquakes, hurricanes, or floods
Hardware	Hardware component	Failures of cable
		equipment, power supplies, or facility components, dock or clock synchronization failures
Software	Software	Software; under normal operation in software, so, software recovery mode
Overloads	Service demand exceeds the designed system capacity	
Vandalism	Sabotage or other intentional damage	

Table 4.1

4.2 Findings

Figure 4.2 summarizes the number and duration of outages, customers affected, and customer minutes by cause. Figure 4.3 shows the percentage of outage minutes for each major category. Figure 4.4 shows the percentage of outage minutes by duration. The data show that the duration and magnitude of outages differs significantly for most failure categories. For example, although overloads caused only 6 percent of the total outages, they accounted for nearly half the total customer minutes. Human error caused nearly half of the outages, but only about a quarter of the downtime.

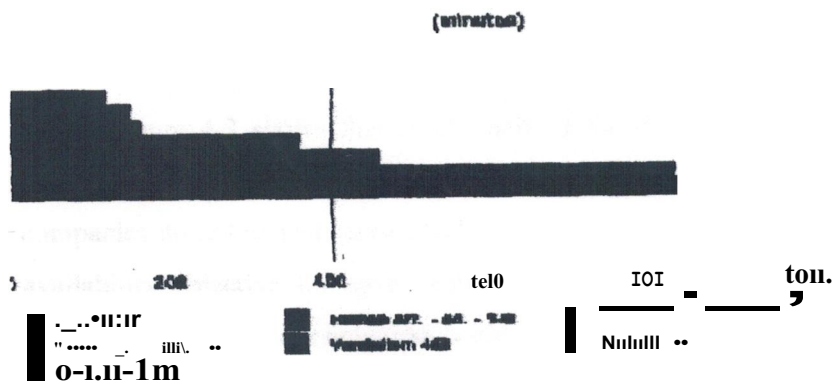


Figure4.2

Figure 4.2 illustrates the outage durations for the different failure categories and reveals part of the reason number and magnitude measures differ. Software, hardware; and human error by company personnel caused the shortest duration outages. Figure 4 compares the duration and customers affected for the major failure categories. The x axis displays outage duration, while the y axis displays the number of customers affected. Only overloads and acts of nature (in the upper right corner) are extended and widespread. Failures due to the errors of telephone company personnel (upper left) are brief but have widespread effects. Hardware and software failures were similar in terms of outage duration and customers affected. Vandalism and human errors caused by others were also similar in their effects. Table 4.2 Failure effects by categories and sources, for outages from April 1992 to March 1994

Categories and Sources outages	Average no, of Average Customer		
	No.of outages	customers	Customer
		affected	minutes
		Affected	duration (in

Table4.2

4.3 Observations

Figure 4.3 shows that nearly half of the downtime is caused by (planned) outages, which are expected outages. Because of technical and technical constraints, telephone companies do not expect service to be available all the time. For example, Bell Canada's availability objective for local networks in its client companies is 99.93 percent, lower capacity networks collectively demonstrate a trust of this downtime but increase cost. Through decades of experience the telephone industry has established a balance between benefits and the cost their consumers find acceptable.

Although the errors attributed to telephone employees are not the major source of outages, they are the major source of failure in the operation of aspects under the companies' control. Human error by company personnel accounted for only 25 percent of outages and 54 percent of downtime. But failure is controllable by the telephone companies (human error plus hardware and software failures) accounted for 56 percent of outages and 21 percent of downtime. So human errors by employees contributed nearly half of these outages (25 divided by 58) and nearly two-thirds of customer minutes of downtime (14 divided by 23). Effects of human error were about the same for hardware and software maintenance. Human error for maintenance of cable and hardware equipment accounted for about 15 percent of outages and 7 percent of downtime. Software-related human errors included mismatched versions, incorrect data entry, and procedural errors during upgrades. These errors accounted for 11 percent of outages and 17 percent of downtime.



Figure 43

Software errors caused a significant number of moderate outages. Although software errors caused approximately 14 percent of the outages, they accounted for only 2 percent of the customer minutes excluding human error by others, acts of nature; and overloads; however, software errors accounted for 24 percent of outages and 9 percent of customer minutes (dwntime). Two factors probably cause software outages to be the worst of human intervention capabilities in the VSTN and the use of extensive error detection and recovery software.

4.A. Why So Reliable?

Despite its enormous size and complexity, the PSTN manages an availability rate better than 99.999 percent in the time period studied. Why should perhaps the world's largest and most complex computerized distributed system also be among the most reliable?

4.1 Reliable Software

To begin with, telephone switch manufacturers are among the world's leaders in computer technology. They focus much of their research on developing highly reliable systems. Their software development processes typically include the most sophisticated practices, supplemented by elaborate quality assurance functions. The

software's focus demonstrates that we can develop highly reliable software using the best practices.

4.4.2 Dynamic Rerouting

But other factors add to the PSTN dependability. In particular, telephone network designers appear to have exploited some aspects of the network's nature to:

By its very nature, the telephone network is highly distributed; standardized failures are more likely, and switches can reroute traffic dynamically to avoid a failed network node. More important, intermittent failures are usually not catastrophic. Otherwise, systems face much greater risks from a failure, no matter how brief. For example, failure of a few seconds in some military, wire avionics software may result in the aircraft's destruction. A brief failure in one network component has relatively little impact on the availability of the entire JSTN across the US. However for the PSTN to remain, it must keep a good deal of information globally. Maintaining consistent distributed databases is a very complex, in itself, a system components.

In his book *Normal Accidents*, Charles Perrow identified two factors: "interactions and coupling" that are significant in determining a system's safety properties. Perrow's terms refer to the dependencies between components. "white coupling" refers to the flexibility in a system. He characterizes it as linear, or, complex, while, coupling, is loose or tight. Systems with simple, linear components have components that affect only other components that we functionally observe. Complex systems entail interaction with many other components in different parts of the system. Loosely coupled systems have more flexibility in time constraints, operation sequencing; and assumptions about the environment than do tightly coupled systems, with components interacting in a right manner, are likely to provide a better result. Complex interactions allow for more complicated development and make the system harder to understand and predict. Tight coupling also means that the system has less flexibility in recovering when things go wrong.

John Rushby applied Perrow's, *analysis* of failures in large physical systems to computer systems [7]. In such systems, interactions can, for example, take the form of

signaling lines coordinates processes or keeps distributed databases consistent. Coupling refers to the ordering of operations, acceptable input data ranges, and the flow of data through the system. For example, the Cofitro systems, which implement real-time control of a refinery, are tightly coupled, whereas the Internet, with multiple paths to route packets, is a loosely-coupled example. Systems that require frequent updating of a distributed database are likely to have a message-exchange mechanism for reporting errors and maintain the database's global consistency. A simple, update and reporting system, which updates a database and writes files for input to report programs, is an example of linear interaction.

4.5 Loose Coupling

In most systems, a trade-off can be made between simplicity of interaction and looseness of coupling. We can consider the PSTN a loosely coupled system because it can dynamically reroute calls along many paths. However, it achieves this loose coupling at the cost of some complex interactions between components. These include the need for a knowledge base, interactions among many systems, and the maintenance of some globally consistent databases. Major switching units store information on alternative paths and exchange data on traffic patterns and switch status throughout the day. Such complex interactions can contribute to failures, making system behavior difficult to analyze.

The most spectacular example of a failure due to complex interactions in the PSTN is the nationwide AT&T Retworing failure. This failure resulted in interactions between systems attempting to maintain consistent information about a failed switch. On the other hand, the PSTN distributed database of routing information promotes loose coupling, which, in turn, contributes to system dependability.

For a communications system, coupling is probably the most important of the two properties in determining its capacity to tolerate failures. It is directly related to the system's primary function; maintaining connections between points. The PSTN is designed for reliability, and for this reason, it is more complex than the PSN. Loose coupling probably more than makes up for the increased complexity. Designers must consider the trade-off

Between these factors, the inherent interaction is a factor in the design of the system.

high-integrity system. Two levels of recovery mechanism: automated and manual.
 • explanation: PET - legacy - code: tplh'g,

Designers, devote about half of the software in telephone switches to either detection or correction. Such a large percentage of self-checking is: probably atypical for software systems. Although some researchers note that adding fault-tolerance and fault-avoidance mechanisms to software sometimes decreases dependability' because of the memory management complexity; these mechanisms work with great success in switching systems. Older computer-driven systems might benefit from more: extensive use of built-in diagnostic and recovery software.

4.6 Human Intervention

In addition to built-in self-test and recovery mechanisms, operators monitor telephone switches. 24 hours a day and usually have the ability to switch software, via a fly-by wire system, via a terminal; provide 24-hour support services; \$W~y with a remote maintenance capability that allows them to service software, in a switch thousands of miles away. Human intervention corrected many failures in less than one hour. Simply "restarting" a switch temporarily fixed a significant number of software-related outages.

Through monitoring benefits from automated human operations. Using information on switch status and traffic patterns exchanged by switches, software within a switch will automatically select an alternative route if the preferred route becomes overloaded or unavailable. In the Switching and Control Systems (SCS) program, Intervention can reconfigure the network, sometimes solving the problem in a few minutes. Status data exchanged regularly between switches makes automated and human operations, to reconfigure, routing, possible. PSFV designer, the cepting-interactions - time - in favor of human: computer LaO'Se tr('ftp'g; aUg;w...s,, mnn,,tn operators to intervene in the event of failure, rather than relying entirely on computer control.

Software is not the weak link in the PSTN system's reliability. Extensive use of built-in self-test and recovery mechanisms in major system components (switches) contributed to software dependability and are significant design features in the PSTN. The network's high dependability indicates that the trade-off between dependability gains and complexity

introduced by built-in self-test and recovery mechanisms can be positive. Likewise, the tradeoff between complex interrelationships and the reliability of system components has been positive, permitting quick human intervention in most system failures and resulting in an extremely reliable system.

CONCLUSION

Despite its enormous size and complexity, the PSTN averaged an availability rate better than 99.999 percent in the time period studied. Why should perhaps the world's largest and most complex computerized distributed system also be among the most reliable. To begin with, telephone switch manufacturers are among the world's leaders in computing technology [5]. They focus much of their research on developing highly reliable systems. Their software development processes typically incorporate the most sophisticated practices, supplemented by elaborate quality assurance functions. The PSTN software's low failure rate demonstrates that we can develop highly reliable software using the best practices.

A circuit switched network, such as the PSTN, provide end to end connection on demand, as long as the necessary network resources are available the connection's end to end delay is usually small and always constant, and other user's cannot interfere with the quality of communication.

Until 1981 the MCI Mail system was the only commercial electronic mail system. It was a circuit switched network, and it provided a reliable means of communication. The impact of the invention was much wider as it provided the basis for the current telephone network.

In 1901 Guglielmo Marconi demonstrated that radio waves could be used to transmit information over long distances when he sent a radio message across the Atlantic Ocean. This enabled the electronics revolution to take place and provided the basis for a computerized, rather than mechanical, telecommunications network.

In 1947 William Shockley, John Bardeen, and Walter Brattain invented the transistor. This enabled the electronics revolution to take place and provided the basis for a computerized, rather than mechanical, telecommunications network.

The modern telephone network can be viewed as a globally distributed machine that operates as a single resource. Much of it uses interconnected computers. The network, that most people use to carry voice traffic, can also be used to transfer data in the form of pictures, text, and video images.

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