Chapter Contents

2.0 Telephone System

Objectives

In this section we will examine:

- The various steps needed to make a telephone call
- The various types of telephone offices and how they are organized
- How telephone exchanges communicate with each other
- How a class 5 end-office is organized

2.1 Overview

☞ Fundamentals of Telecommunications

From the western perspective, the telephone is ubiquitous. However, the reality is that 80% of the world's population has limited access to one.[1]

To the average person, the telephone system is simply a 'black box'. It provides dial tone, accepts telephone numbers, makes connections, rings phones and keeps track of billing. Relatively few people need to have any idea of how the PSTN works. The main consideration is that it works when needed, and is relatively inexpensive.

In the early days of telephony, an operator was needed to make connections between subscribers, but as technology improved, methods were devised to allow the customer to control the switching operation. Telephone numbers were assigned to each customer, and rotary dials that converted numbers to dial pulses were developed. Automated switches used these pulses to activate relays, and ratchet mechanisms to make the desired connection. However, more was needed than just hardware, a completely new protocol or customer procedure had to be developed.

2.1.1 A Telephone Call

To make a telephone call, one simply picks up the handset, dials, and waits for the system to perform its magic.

Lifting the handset from its cradle releases a hook switch and causes a current to flow. The central office monitors this loop current and interprets an off hook condition as a request for service. .

The office acknowledges the request for service by sending dial tone. This normally occurs in less time than it takes to pickup the handset and place it to the ear. Once dial tone has been received, the subscriber starts to dial.

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1 Communications News, May 1995

When dialing, the rotary dial switch opens and closes the loop in a predetermined manner. If one was very coordinated, it is possible to perform the same task by flashing the hook switch. To assure the customer that the system is responding to the request for service, dial tone is removed from the loop once dialing starts.

Standard Touch Tone Telephone[2]

http://howstuffworks.com/telephone.htm

Depending upon the type of office and digits received, a number of things might happen. In most cases, end-users are attached to what is called a class 5 or end-office. These are the most common types of telephone exchanges. Each class 5 office has one or more, three digit exchange numbers. These are the first three digits in an ordinary 7-digit telephone number

If the central office includes the customer dialed exchange number, it will know that the call is local and the other party is connected to the same office. The office will therefor control the entire call setup and takedown.

If the first three digits do not correspond to an exchange handled by the end-office, it will have to find a trunk line to an office that can handle the call. This means that each office must know the exchange numbers of all the offices within its calling area, and how to get to them. The call setup and takedown will therefor be shared between the two exchanges. They must monitor the call in progress and inform each other of any change in call status.

If the first digit dialed is a one, the office will recognize this as a long distance call, and will start looking for a spare toll trunk. A toll office has a greater knowledge base as to where distant exchanges are located and how to get to them.

The telephone system must be intelligent enough to recognize that in a local call, only seven digits are usually required. Some exchanges however, allow local calls by omitting the exchange number and using only the last 4 digits or extension number. In large urban areas, it may be necessary to prefix local calls with a 3-digit area code whereas, in other less dense urban areas, a local call can be made to a different area code but without the need for dialing the code. An international call may require up to 16 digits.

Once the entire number has been received, the office at each end of the connection must alert both parties as to what is happening. At the originating end, a ringing tone is sent to sound the speaker in the handset. At the terminating end, the office is generating a much larger ringing voltage to activate a bell.

The far-end-office monitors the line to determine if someone answers the ringing phone. This is done by examining the DC current drawn when the far-end customer lifts the handset, inducing loop current through the hook switch. The

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2 Whith D. Reeve, Subscriber loop Signaling and Transmission Handbook, figure 1-6

far-end-office must then disconnect the ringing before the handset reaches the ear, and signal back to the originating office that someone has picked up the phone. The origination office must then disconnect the ring back tone and complete the voice connection.

Both end-offices monitor their respective loop currents during the entire call to determine if one party hangs up. Once this happens, one end-office signals the other, and dial tone is placed on the loop. This alerts the remaining party that the connection has been terminated.

If the line is occupied when a call is attempted, the central office will not set up the connection and return a busy tone to the originator. By doing this, switching, call processing, and transmission resources are not being tied up unnecessarily. However, there are a number of options such as call forwarding and call waiting which modify this process.

With call forwarding, a call to a busy number is routed to an answering service. With call waiting, the calling party hears a ringing tone, and the called party hears a beep, which they can either ignore or signal back to the office that the new call should be given priority over the existing call. If however, the call cannot be completed because the system itself is too busy, it returns a fast busy tone to the originator.

In a touch-tone environment, the same procedure is followed, except that tones are used to convey numbers to the local office instead of interrupting loop current. Some calling features, generally known as CLASS[†], are available only in areas with touch-tone service.

Electronic Telephone with Microprocessor Interface[3]

2.1.2 But where do the telephone wires go?

The telephone line goes to a terminal block in a service area interface. These are often located on a pole or small enclosure on the street. The service area interface bundles the subscriber drop cables into a single larger cable. These are in turn gathered together to form larger feeder cables. The entire wiring system somewhat resembles a huge tree.

Cables coming out of a central office may have hundreds or even thousands of pairs bundled together however by the time the cable gets to the end user, it is generally down to about 50 pairs. An individual subscriber consists of many cable sections spliced together. Bellcore claims that the average U.S. subscriber line has twenty-two splices.

Feeder cables enter the central office in a large underground room called a cable vault. Each feeder may contain hundreds of pairs of wires, and be pressurized in order to prevent moisture or ground water from entering and affecting the

† Custom Local Area Signaling Services

3 Motorola Communications Device Data Book

transmission characteristics of the wire. A typical vault may contain tens of thousands of wire pairs, ultimately going to end-users.

The cables pass through the vault and are terminated on the vertical side of the MDF[†]. To protect the central office equipment from high voltage transients, such as lightning strikes, which may travel down the wire, the lines are surge protected by carbon blocks or gas tubes.

The horizontal side of the MDF, connects the incoming telephone lines to the peripheral equipment. All that is required to connect a line appearance to a specific interface is to place a jumper between the vertical and horizontal sides of the MDF.

Signals coming from an end-user are generally analog in nature. Consequently, the peripheral equipment converts the signals to digital form before passing them on to the rest of the network.

Incoming trunks from other central offices are comprised of specialized carrier systems. They may be either analog or digital, but all new systems are strictly digital.

Most end-user voice & data interfaces are multiplexed on to high-speed paths, which pass through the internal switching, network before being routed to outgoing lines or trunks. Incoming digital carrier systems may be accepted directly into the switching network through a cross-connect or may be demultiplexed prior to switching.

2.2 PSTN Hierarchy

Box Telephone 1877

Telephone Office 1888

Historically the telephone network was composed of a hierarchical structure consisting of 5 different office types. The most common of these is the class 5 end-office. An end-office connects directly to subscriber telephone sets and performs switching functions over a relatively small area. Telephone exchanges connect to subscribers by means of local loops or lines, generally one per customer. Telephone offices connect to each other by means of trunks.

Class	Office Type	Approx. number [US][4]
1	Regional Center	10
2	Sectional Center	67
3	Primary Center	230
4	Toll Center	1300
5	End-office	19000

† Main Distribution Frame

4 *Data and Computer Communications ;* Stallings

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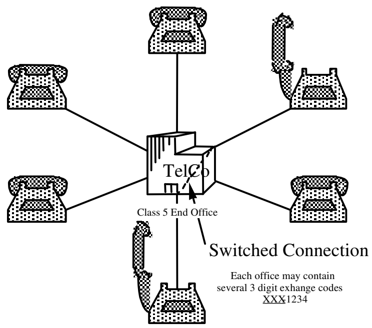
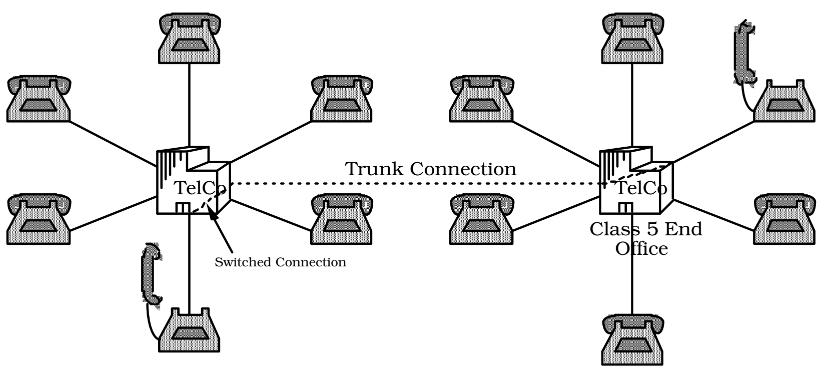
2.2.1 Local Network



A class 5 or end-office interconnects telephones throughout a small service area. Each end-office may contain several three-digit exchange numbers and is aware of other local exchange numbers held by other offices.

Calls between offices are routed over interoffice or tandem trunks. Long distance calls are routed to toll offices via toll trunks. The average class 5 office serves approximately 41,000 subscribers, and covers 30 square km in an urban environment.



Some nodes may have no customers at all, and may be connected only to other nodes. These inter-node or trunk connections are usually made by FDM† or TDM transmission links.

2.2.2 Exchange Area Network

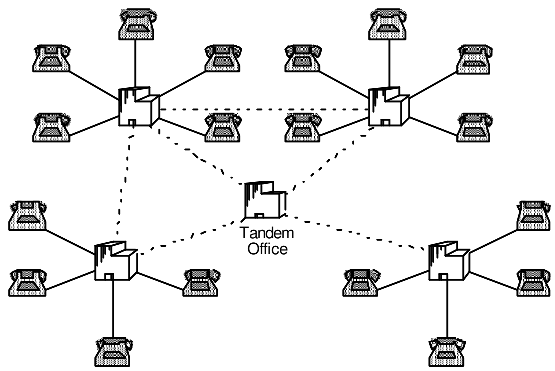
An exchange network consists of local and tandem exchanges connected by trunks. A tandem office interconnects class 5 offices by means of twisted pair, coax, microwave, or fiber optic carriers. Alternate routing paths between local exchanges are provided if the direct trunks are occupied.

An exchange area includes all of the offices, which are aware of each other, but do not involve long distance charges. In very large urban areas, there is an

† Frequency Division Multiplexing

overlap between exchange areas, which may also cross over area code boundaries.

2.2.3 Long Haul Network



A long haul network consists of exchanges interconnected by toll offices. Toll offices keep track of long distance charges and are typically confined to national boundaries. These trunks consist of high capacity coax, microwave, or glass fiber.

Messages used to control the call setup and takedown can be sent by two basic methods. Traditionally, inter-office messages are sent over the same channel that will carry the voice path, but in newer systems, common channel signaling is being employed. In this method, the offices have dedicated facilities, which are used to send inter-office messages. There are some advantages to this, perhaps the notable being the added degree of difficulty encountered if one wants to defraud the system. When in-band signaling was used, it was possible for people to dial long distance calls without being charged, if they created the tones used to disable the toll circuit.

2.3 Interoffice Signaling

Trunks are used to interconnect the various levels of telephone exchanges. It is necessary for these links to exchange on a wide range of information including:

- Call related signaling messages
- Billing information
- Routing and flow control signals
- Maintenance test signals

_____ Notes _____

There are two ways for telephone offices to communicate with each other and pass on routing information. Information can be conveyed in the same channel that will be used to convey the voice signal, or it may be completely disassociated with it.

2.3.1 CAS[†]

This approach uses the voice channel to send information through a trunk. For example, a 2600 Hz tone is used in interoffice trunks to signal on-hook. A major disadvantage of this system is that subscribers can bypass toll centers by injecting the appropriate tones. One way to avoid this problem is by using out-of-band signals on toll trunks. Since the customer's signal must pass throu gh an audio anti-aliasing filter, it is not possible to inject the out-of-band signaling tone.

A principle advantage of in-channel signaling is that the integrity of the voice path is checked each time a connection is established. Out-of-band signaling allows for continuous supervision of the connection throughout the call.

2.3.2 CCIS[†]

This approach has the signaling information conveyed on a facility completely separated from the customer's voice path. This allows for a faster, more efficient control, however the reliability of the CCS network must be considerably greater than that of the individual voice paths. The signaling channel may follow the same route as the final connection path, or it may be completely disassociated with it. STPs[†] are need in the network if the signaling path is disassociated, thus effectively creating two networks: a speech network and a signaling network overlay.

2.3.3 SS7

☞ SS7 Tutorial by Bell Atlantic

☞ SS7 & Intelligent Networking Applications by Natural MicroSystems

Virtually all calls requiring tandem or toll office routing are established and controlled by the SS7 signaling network.

The SS7 signaling network is a packet switching facility comprised primarily of STP[†]s and SCP[†]s connected to the PSTN (SSP[†]). STPs are deployed in pairs

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† Channel Associated Signaling

† Common Channel Interoffice Signaling

† Switch Transfer Point

† Signaling Transfer Point

† Service Control Point

† Signal Switching Point

and are the brains of the system. They determine which trunks and offices should be used in establishing inter-office connections.

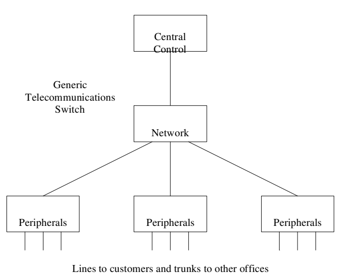
The SCP is a database that keeps track of such things as: credit card authorization, virtual network subscriber listings, 800 number conversion tables, billing, and other special services.

2.4 Class 5 Office

A telephone central office is often referred to as a switch because it switches or routes calls. Regardless of who makes them, all class 5 offices have the same objectives, and therefor have similar structures. The three major components found in any modern switching systems are; the central control, network, and peripherals.

2.4.1 A Generic Communications Switch

The internal architecture of a telecommunications switch is somewhat like the organization of the entire system. The internal structure is often illustrated by the traditional pyramid or hierarchical arrangement. The control or brains of the operation are shown at the top, and the peripheral units that connect to the outside world are placed at the bottom.



The MTBF† for any PSTN switch must be very long, since business would soon grind to a halt if telephone traffic was interrupted for a prolonged period, but more importantly, emergency services would be severely curtailed. For these reasons, large public switches have a great deal of redundancy built in. Redundancy is provided in two basic ways; hot standby and load sharing.

In the hot standby arrangement, two or more processors are fed with the identical information and are making decisions, however, only one of these processors is in charge and is executing decisions. In the event of a failure, the healthy unit assumes the full load. There is no degradation in performance, and no calls in progress should be lost.

In a load-sharing configuration, all processors are actively working but not to their full capacity. In the event of a failure, the defective processor is isolated from the system, and the others pick up the slack. There may be degradation in performance, and calls in progress on the defective processor may be lost.

† Mean Time Between Failure

_____ Notes _____

2.4.2 Peripheral Layer

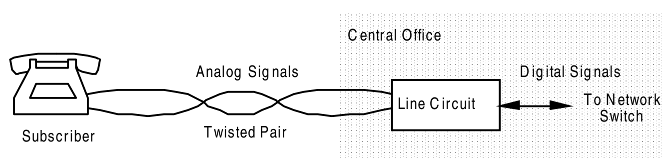
The periphery is the outermost layer in any system. It contains the interface to all outgoing lines and trunks.

2.4.2.1 Line Interface

The line interface is often referred to as a BORSCHT circuit. This acronym describes the functional requirements of a standard telephone line interface. The tip and ring leads of the telephone set are wired through some protection devices to the line interface located in the peripheral module. This interface must perform the following functions:

- B Battery feed
- O Over voltage protection
- R Ringing
- S Supervision & Signaling
- C Coding
- H Hybrid
- T Test

Many of these functions can be integrated into a single IC, often called a SLIC[†] chip. SLICs have been available for the PBX market for over a decade. Recently however, they have also become available for the central office environment as well.



2.4.3 Network Layer

The network switch does the actual routing of signals from one customer or port to another. Switching can be done in either the space or time domain. Initially, all switching was done by mechanical contacts in the space domain. Today most switches are digital in nature and operate mainly in the time domain. However, some degree of space switching is always required since signals must ultimately be routed from one line to another.

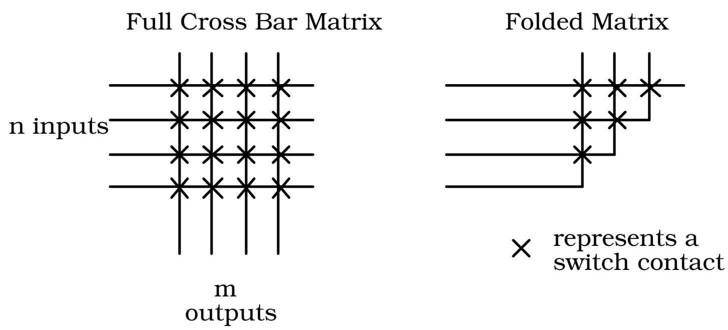
Networks always provide some form of concentration. That is to say, not all customers can be handled simultaneously. Statistical analysis shows that in the majority of cases, the switch needs only to handle about 20% of all the subscribers at one time. This is similar to highway systems where the roads are designed to handle not all vehicles simultaneously, but only a certain peak load.

† Subscriber Line Interface Circuit

2.4.3.1 Space Division Switching

The physical path between any two customers on a space switch is not shared with anyone else. Crosspoints made from electromechanical relays have been used to perform the interconnection, but newer systems use semiconductors.

In a crossbar matrix, the number of inputs and outputs do not have to be equal thus facilitating either concentration or expansion. In any case, a total of *N*x*M* crosspoints are required. Although only one contact is shown, many systems require two contacts if they keep the integrity of the tip and ring leads throughout the system.



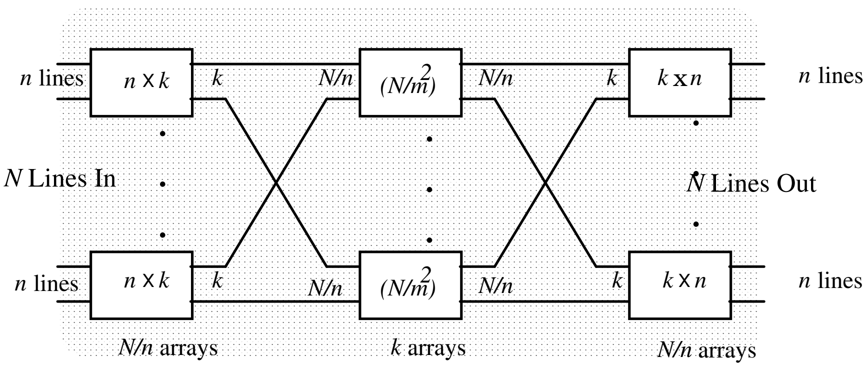
The folded matrix treats inputs and outputs identically and requires only $N(N-1)/2$ crosspoints.

The major disadvantage of this type of switch is the rapid increase of crosspoints as the system size increases. The number of crosspoints increases as N^2 and yet only a maximum of N crosspoints can be active at any given time. Failure of a single crosspoint prevents communication between the two devices sharing that crosspoint.

2.4.3.2 Multiple Stage Space Switch

One way to avoid the cost penalties associated with full matrices, is to organize the contacts into smaller groups. This impacts the call processing since the switch controller must manage several contacts per connection. Furthermore, the connection between any two subscribers may take any one of a number of paths, thus further complicating the decision making process.

The following sketch shows a simple three-stage space switch. It should be remembered that since a full duplex connection is required, a second switch supplying the return path must be provided.



3 Stage Space Switch

The total number of crosspoints in the above illustration is:

$$2\left[\frac{N}{n}\text{ arrays}\times(n\times k)\frac{\text{crosspoints}}{\text{array}}\right]+k\text{ arrays}\times\left(\frac{N}{n}\right)^2\frac{\text{crosspoints}}{\text{array}}$$

or

$$2Nk+k\left(\frac{N}{n}\right)^2$$

The number of crosspoints required is dramatically reduced in this multiple stage switch in comparison to a square matrix. The central stage allows several ways to make a connection between two subscribers therefor, single crosspoint failures can be bypassed. This results in a more flexible and reliable system, but demands a more complex control structure.

Blocking

A non-blocking network is capable of finding a path between any idle input to any idle output. This does not mean that the system be able to handle simultaneously all customer requests for service. In such a case, the system may overload but the customer is not able to distinguish the difference between blocking and overload.

The following illustration shows how blocking can occur. The solid lines represent connections in service.

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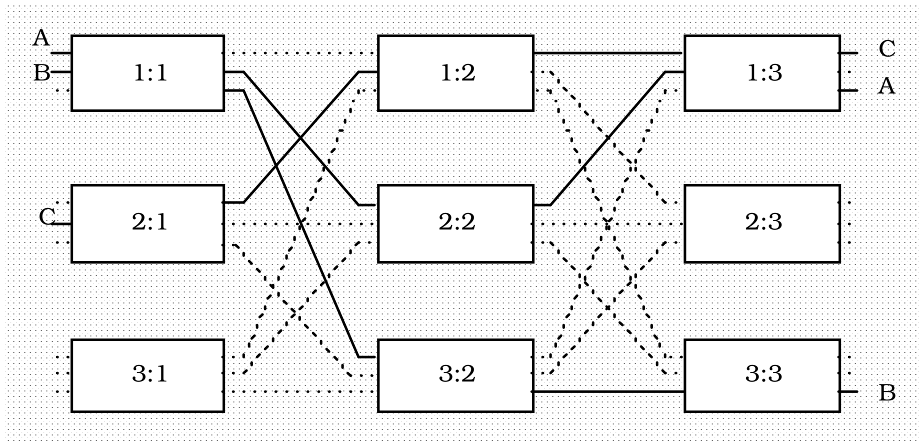


An Example of Blocking

The connections in service are not necessarily the optimum routing and may have been forced by the prior connections. Note that in this case, it is not possible for the last customer on switch 1:1 to contact the last customer on switch 1:3 because there is not a free center switching stage common to both. To overcome this, an additional center stage can be added. To prevent blocking, *2n-1* center arrays are necessary.

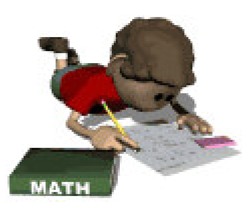
The total number of crosspoints (*NoC*) in a non-blocking network is therefore:

$$NoC = 2N(2n-1) + (2n-1)\left(\frac{N}{n}\right)^2$$

$$N = total\ number\ of\ lines\ in$$

$$n = number\ of\ lines\ in\ each\ input\ stage$$

As the total number of lines in increase, the total number of crosspoints can be approximated by: $NoC \approx 4N\left(\sqrt{2N} - 1\right)$

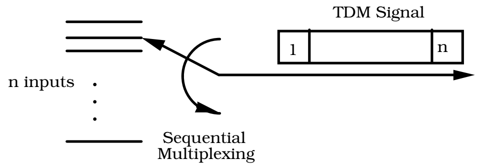


A non-blocking network of this type is known as a Clos switch, after its inventor. Its basic characteristics include:

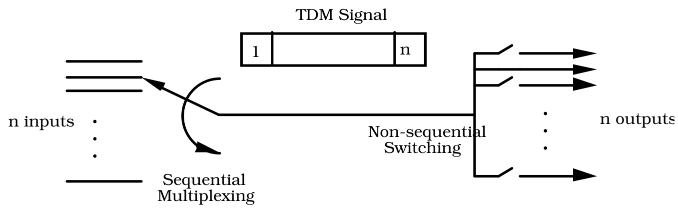
- Expansion in the first switching stage
- An odd number of center stages
- Concentration in the last stage

2.4.3.3 Time Division Switching

Time domain switching is simply an application of time domain multiplexing and may be performed on analog or digital signals. Any number of inputs may be sequentially routed to a single output.



This technique increases the transmission link utilization and can be modified to support circuit switching. If a multiplexer is placed at the input, a demultiplexer is placed at the output. This system can be used to multiplex either analog or digital signals.



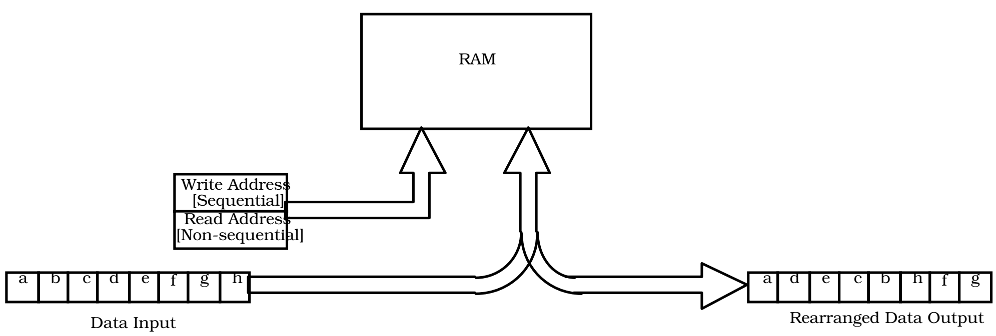
TDM Bus Switching

Each customer is assigned a unique switch, but all customers share the same internal signal path. For *N* customers there are *2N* switches. TDM bus switching occurs when the input sequence is not the same as the output sequence.

Each customer is given access to the common structure for a brief moment. If higher data rates are needed, multiple inputs can be assigned, thus giving the customer more time to transmit a signal.

Time Slot Interchange [TSI]

If information can be arranged into a sequence, it can also be rearranged, much like shuffling a deck of cards. This is the task of the time slot interchange unit. Full duplex operation is achieved by interchanging time slots in both directions.



The incoming TDM channels are mapped sequentially into RAM while the outgoing channels are read out non-sequentially. This output address generator is simply a memory-mapped pointer governed by a central controller.

The required memory access time can be approximated as the inverse of the channel rate. The RAM width is determined by the number of bits in a channel, and the length by the number of channels in a frame.

Example:

For a T1 link (24 channel PCM), there are:

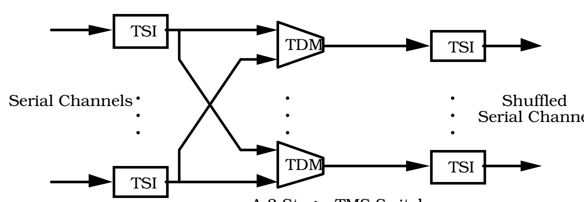
$$24\,\frac{\text{time slots}}{\text{frame}} \times 8000\,\frac{\text{frames}}{\text{sec}} = 192{,}000\,\frac{\text{time slots}}{\text{sec}}$$

Therefor the maximum read/write cycle time is:

$$\frac{1}{2 \times 192{,}000} = 2.6\,\mu\text{Sec}$$

Time Multiplexed Switching

Combining TDM and TSI allows a channel from one digital bit stream to be switched to any channel on another digital bit stream. A multistage time switch can consist of cascaded switching modules. To prevent blocking, 3 or more stages are required.



A 3 Stage TMS Switch

Although it appears that only time domain switching is used in this example, it is also known as a Time-Space-Time switch or simply *TST*. This is because the center stage is actually switching different input lines in space to a common output line. Since there is an ambiguity in the terminology, some manufacturers of telecommunications equipment may refer to this as a *TTT* switch if all of the signals are digital.

For small-scale switches, space switching is most efficient. However, as switch size increases, time domain switching tends to gain the advantage.

It is sometimes difficult to make direct comparisons between various telecommunications switches because the internal architectures may be quite different. However, it is possible to compare BHCA† capacity or performance such as traffic intensity, under a specified set of circumstances.

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† Busy Hour Call Attempts

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Traffic Intensity the product of average holding time and the calling rate and is expressed in CCS or Erlangs.

CCS† is defined as:

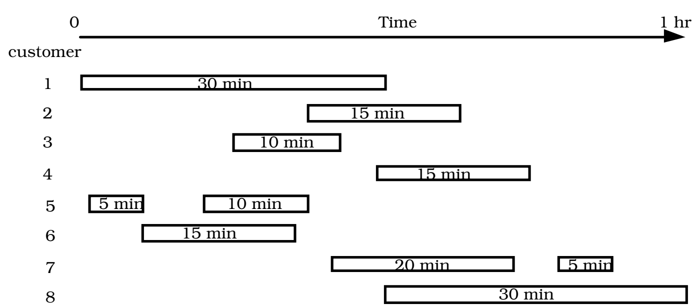
$$CCS = \frac{\#\text{calls per hour} \times \text{call holding time in seconds}}{100 \text{ seconds}}$$

Therefore, 36 CCS = 1 Erlang

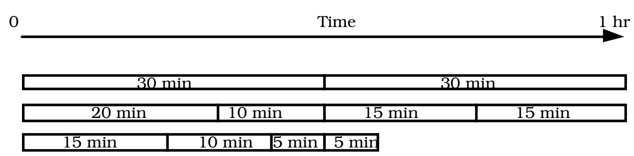
An erlang is a measure of the traffic intensity and is equal to the average number of simultaneous calls at any given moment. It represents the total circuit usage during any time interval, divided by that interval. It also corresponds to the minimum number of channels necessary to carry the traffic, if it could be scheduled.

Example

Imagine for a moment that there are 8 customers in a small telephone system making random calls:



These calls could conceivably be arranged as:



Three, 1 hour channels could carry this traffic and still have 25 minutes left over to spare. The traffic intensity is therefor less than 3 Erlangs:

$$\text{Traffic} = \frac{155 \text{ minutes used}}{60 \text{ minutes period}} = 2.58 \text{ Erlangs}$$

or in term of CCS:

$$\frac{10 \text{ calls per hour} \times \text{average of } 930 \text{ seconds per call}}{100 \text{ seconds}} = 93 \text{ CCS}$$

† 100 Call Seconds

Example

Given that the average calling rate during the busy hour in a 10 K line office is 2.1 calls per line, and the average call duration is 3 minutes 12 seconds, find the traffic load and intensity:

The total traffic load is: 10,000 x 2.1 = 21,000 BHC

The traffic intensity is:

$$21,000\ \text{BHC} \times \frac{192\ \text{seconds per call}}{100\ \text{seconds}} = 40{,}320\ \text{CCS}$$

or

$$21,000\ BHC \times \frac{3.2\ \text{minutes per call}}{60\ \text{minute interval}} = 1,120\ \text{Erlangs}$$

note that 1,120 x 36 = 40,320

The actual CCS per line as specified in industry is actually much more complicated than this simple description indicates. Other factors, which must be considered, are: interoffice calls, abandoned calls, business or residential mix, etc.

2.4.4 Central Control

This contains the system intelligence and customer database. It knows who the customers are, what they want, and how to provide the service they require.

In a step x step [step by step] switch, the intelligence is fully distributed and there is no central control, whereas in a crossbar facility, all of the intelligence is resident in a central controller or computer. In all modern systems the intelligence is somewhat distributed, with various functional blocks contributing to the decision making process.

At onetime there was a sharp distinction between computers and telecom switches, but today this division is less clear, and central controllers may be regarded as a specialized computer.

Additional Material for Advanced Studies

- Understanding Voice Technology by Motorola
- http://www.dcbnet.com/
- http://www.alliancedatacom.com/
- http://dmsweb.badm.sc.edu/grover/telecom/
- http://www.devrytcom.com/TCM250accel.html

Deployment of Telecommunications Networks by Expertech

Emerging Multiservice Network Architecture by Cisco Systems

Next Generation Networks by Taqua Systems

Network CTI Delivering Intelligent Network Services

Numbering Plans require international cooperation. For more information, refer to:

http://www.nanpa.com/

Canadian Area Code Distribution

http://geography.about.com/science/geography/msub70.htm?iam=mt&terms=%22north+american+numbering+plan%22

2.2.4 DNHR Network†

http://www.ece.arizona.edu/~trangmoe/dyroute/dyroute.html

http://www.cs.cornell.edu/skeshav/book/slides/routing/ppframe.htm

The older, rigid form of office classification is gradually being changed as more powerful switches are brought into the system. Traditionally, the class one office was the international gateway, and the class 5 office attached to the end user.

† Dynamic Non-Hierarchy Routing Network

_____ Notes _____

Although there is still a separation between gateways and end-offices, new telecommunications switches can be configured as multiple office types.

In the traditional system, a high-level switch failure tended to isolate large sections of network. To alleviate this problem, DNHR offices are given equal powers to make routing decisions, and have more interconnection options. In effect, the network begins to look like a gigantic load sharing facility.

Network management centers are needed to monitor the system and pass on routing and traffic status information. Below the DNHR layer, the offices are arranged in the typical hierarchical fashion.

2.3.3 SS7

SS7 by Illuminet

Competitive PSTN WAN Access using SS7 by Cronus

2.3.4 IN

The Intelligent Network Tutorial by Bell-Atlantic

The International IN by Telecordia

Intelligent Network by Telecoria Technologies

IN Service Creation by Marconi

The intelligent network increases the capability of the PSTN by supporting SS7 and intelligent peripherals. These devices include interactive voice response systems, advanced call centers, SS7/IP gateways, voice and fax storage.

2.4.2.2 Battery Feed

Most domestic appliances are powered from an electric utility grid. The notable exception to this is the telephone. This is because the telephone should still operate in the event of a power failure. Indeed, the telephone is vital in case of disaster or emergency.

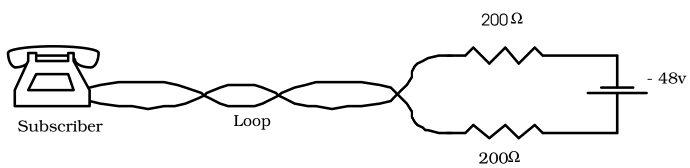
The telephone office provides a nominal -48 volt dc feed to power the phone. This magnitude is considered the maximum safe dc operating potential. It would not be in the telephone company's best interest to provide a dc voltage, which could electrocute its customers, or it's own employees. A negative potential was chosen to reduce corrosive action on buried cables.

Newer multi-function telephones cannot always be powered from the telephone exchange and often require an alternate power source. For this reason, very sophisticated line interfaces such as ISDN SAA interfaces have a ‘fail to POTS’ mode. If the electric power fails, the complex phone cannot function to full capacity. The telephone exchange can sense the local power outage through the telephone loop and switches to POTS only service.

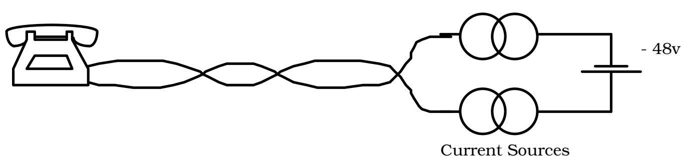
This allows full feature sets to still provide emergency phone service in the event of disaster. To take advantage of this feature requires that the set designers be well versed in central office line interface design.

The POTS loop requires a nominal -48 v at 20 – 100 ma dc to maintain a voice and signaling path. The earpiece in the handset does not require biasing, but the carbon microphone does. Signaling is performed by temporarily placing a short circuit on the loop thus changing the loop current, which is then sensed at the central office.

There are several ways to provide loop current, the simplest being a resistor in series with a battery.



Another way to provide loop current is by an electronic current source:



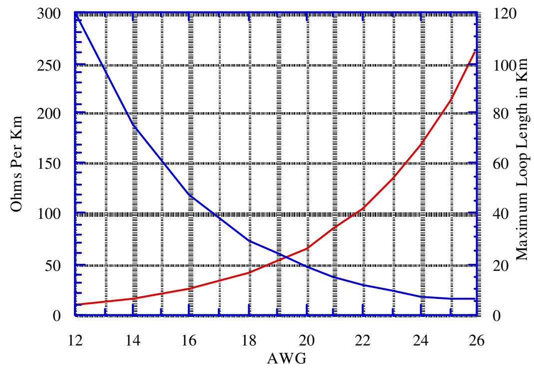
Although this method is quite complex, it has become quite popular with the advent of high voltage bipolar technology. One of the more difficult requirements to meet is the 60-dB longitudinal line balance. To achieve this objective, the impedance to ground on each side of the loop, match within 0.1%. This is easy to do with laser trimmed thick film resistors, but a bit tricky with current sources.

Motorola MC145500 Family Codecs

Motorola MC145554 Family Codecs

About 20 ma is needed to keep a telephone operating. This means that the maximum loop resistance permissible would be about 2000 Ω. In actual practice, the loop is generally limited to 1250 Ω. The maximum loop length is therefore determined by the wire gauge:

_____ Notes _____



2.4.2.3 Over-Voltage Protection

The two major types of over-voltage that can occur are lightning strikes and power line contact. In both situations, the circuit must either recover or fail-safe. Under no circumstances can a surge be allowed to propagate further into the system, or create a fire.

Initial surge protection is provided at the MDF by gas tubes and/or carbon blocks, which arc if the applied voltage exceeds a few hundred volts. Since these devices take a finite time to respond, high-speed diodes are also used at the line circuit inputs.

2.4.2.4 Ringing

Ringing is often provided by means of a dedicated ringing generator that is connected onto the loop by means of a relay. It is possible to generate ringing voltages at the line interface if the current generators have a high enough voltage source available to them. Or alternately, a switching converter with step up capability can be place on the interface.

In Canada, the ringing voltage is a nominal 86 Vrms at 20 Hz, with a 2 second on and 4 second off cycle. On rural party lines, ringing codes of long and short rings are often used.

In the U.S. there are a number of fully selective and semi-selective ringing methods used on party lines. One employs different frequencies ranging from about 16 – 66 Hz. In such cases, each telephone ringer is tuned to its own frequency. Other methods use positive and negative battery voltages to apply ringing on either the tip or ring side of the line to ground to select customers.

2.4.2.5 Supervision & Signaling

The central office must supervise the loop in order to identify customer requests for service. A request for service is initiated by going off-hook. This simply draws loop current from the CO.

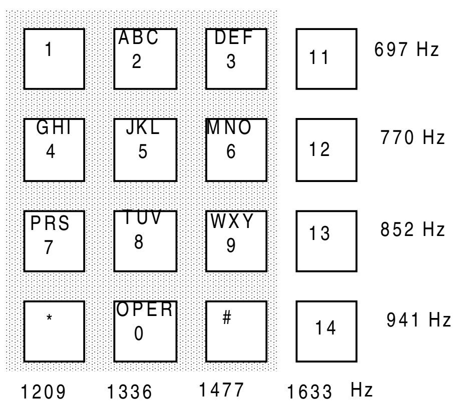
Notes

Loop current, at the far-end is monitored during ringing to enable the CO to disconnect the ringing generator when the phone is answered. The office continues to monitor the loop current at both ends of the connection throughout the call, to determine when the call is terminated by hanging up.

Signaling is a way to inform the CO what we want it to do. The two basic signaling methods used in customer loops are dial pulse and touch-tone. It is interesting to note that preferred customer loop signaling method in analog exchanges is digital, while the preferred method in digital exchanges is analog!

MF Signaling Tones

DTMF Keypad



Two tones are used to perform the signaling function to eliminate the possibility that speech be interpreted as a signal. At one time DTMF decoders were costly and bulky devices located in a common equipment bay, but today with the advent of LSI technology, this function can be performed on a chip. An example is the Mitel MT8865 DTMF filter, and MT8860 DTMF decoder.

Positions 11 to 14 are not presently being used.

2.4.2.6 Coding

The voice coding function is performed by a codec[†]. This device is a specialized A/D and D/A converter, and may include filters and certain control or testing functions.

There are two international standards used to implement coding algorithms today: A-law, which is used in Europe, and µ-Law, which is used in North America. These both involve 8 KHz sampling, and 8 bit A/D & D/A conversion, thus resulting in 64 Kbps digital bit streams. Codecs also compand[†] the signal to keep a relatively constant S/N ratio over the entire amplitude range. Without companding, a 12 bit linear encoding scheme would be needed to obtain the same S/N ratio at low volume levels.

† coder decoder

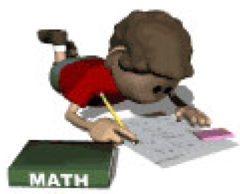
† compress expand

_____ Notes _____

Companding is an operation in which the dynamic range of signals is compressed before transmission and is expanded to the original value at the receiver. This allows signals with a large dynamic range to be transmitted over facilities that have a smaller dynamic range capability. It also reduces the noise and crosstalk levels at the receiver.

μ-Law Algorithm

$$F(x) = sgn(x)\frac{\ln(1+\mu|x|)}{\ln(1+\mu)} \quad 0 \le |x| \le 1$$



A-Law Algorithm

$$F(x) = sgn(x)\left[\frac{1+\log(A|x|)}{1+\log A}\right] \qquad \frac{1}{A} \le |x| \le 1$$

$$= sgn(x)\left[\frac{A|x|}{1+\log A}\right] \qquad 0 \le |x| \le \frac{1}{A}$$

where

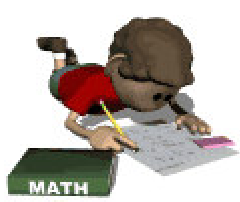
x = input signal

$sgn(x)$ = sign of input (+ or -)

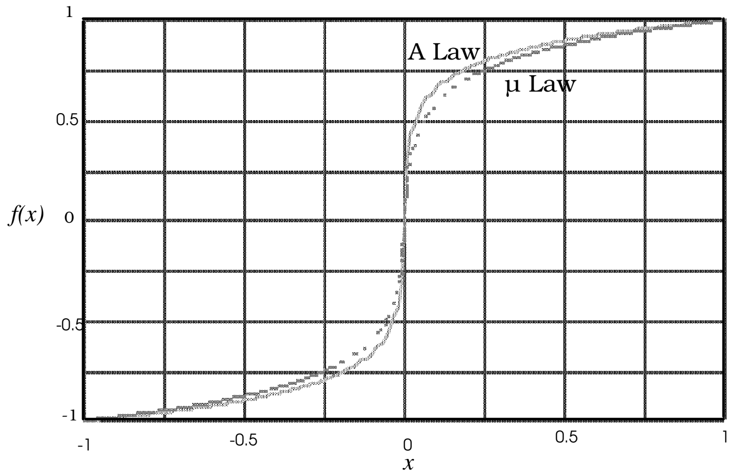
$|x|$ = absolute value (magnitude) of x

μ = 255 (defined by AT & T)

A = 87.6 (defined by CCITT)

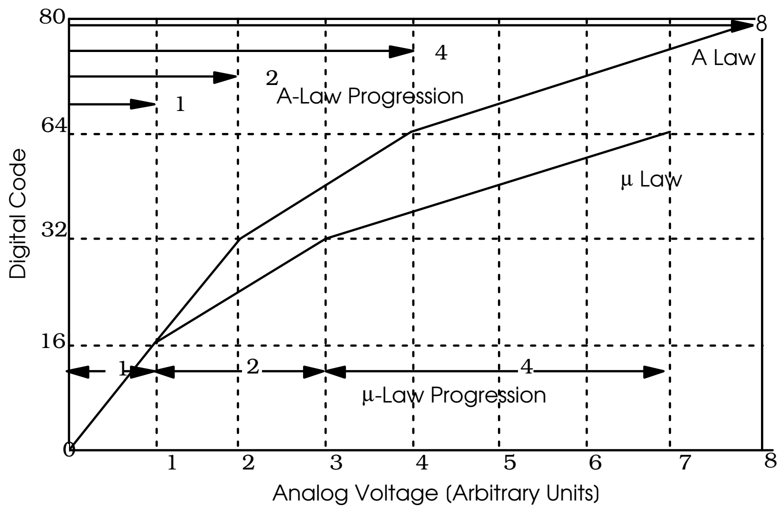


A-Law and μ-Law Curves



Although the actual compression algorithms are continuous functions, the codec approximates them by linear segments. A-law has 13 linear segments, and μ-law has 15 linear segments or cords.

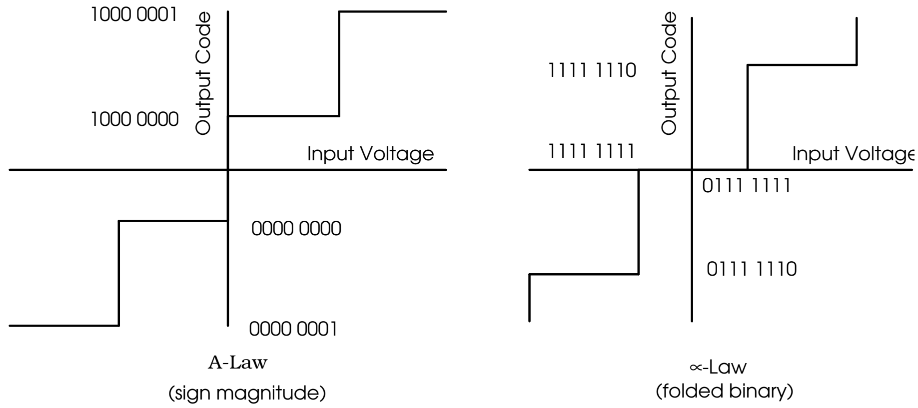
A-Law and μ-Law Segments



Another important difference between the European and North American codecs, can be seen by the position of the decision threshold and its digital value.

_____ Notes _____

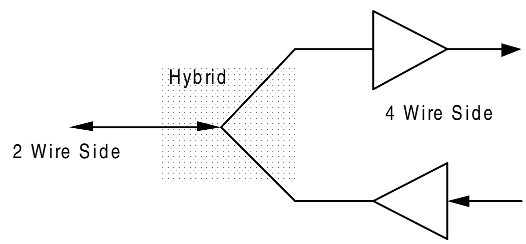
A-Law and µ-Law Steps



When a telephone call is placed between Europe and North America, it is essential that all of these differences be accounted for. It is possible to regenerate the analog voice by passing it through the same type of codec that originally processed it, and then re-code with the other. An alternate approach is to use lookup tables that translate the binary values of one system to the other.

2.4.2.7 Hybrid [or Diplexer]

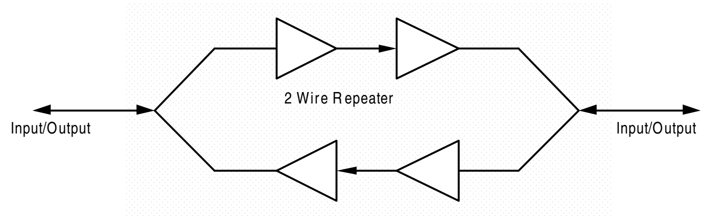
A diplexer performs a bi-directional 2-wire to 4-wire conversion. It allows two unidirectional electrical paths to be combined into a single bi-directional one, and vice versa. It is advantageous to separate transmit and receive portions of the signal since it is easier to make unidirectional amplifiers, filters, and logic devices.



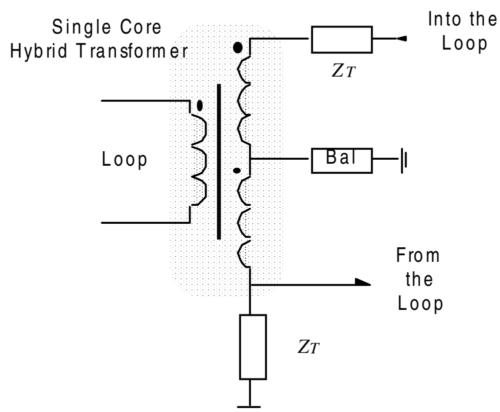
There are bi-directional amplifiers or repeaters that are used in the outside plant environment, but they too use hybrids internally.

Repeaters

By placing two hybrids back to back, it is possible to create a bidirectional amplifier.



The total gain in the 4-wire path within the repeater must not exceed the combined transhybrid loss of the transformers. If this happens, the circuit will oscillate or sing.

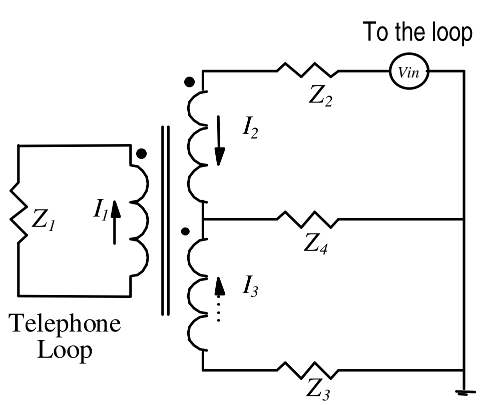


There are several ways to split transmit and receive paths, the simplest method uses a single core hybrid transformer.

The basic defining transformer equations are:

$$\frac{V_1}{n_1} = \frac{V_2}{n_2} = \frac{V_3}{n_3} \quad \ldots \qquad \text{and} \qquad I_1 n_1 + I_2 n_2 + I_3 n_3 \quad \ldots \quad = 0$$

For a single core hybrid with a center-tapped secondary, the impedance relationships for proper operation [conjugate matching] are:



$$\text{for } n_2 = n_3 \qquad Z_2 = Z_3 = 2Z_4 \qquad \text{and} \qquad Z_1 = \left(\frac{n_2}{n_1}\right)^2 Z_4$$

Note what happens if the transformer is driven from one of the secondary windings:

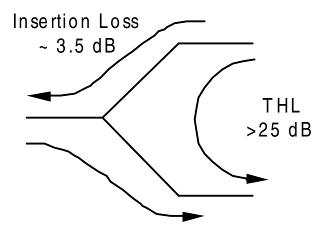
$$\text{let} \qquad n_1 = n_2 = n_3 \qquad\qquad \therefore I_3 = -I_1 - I_2$$

But I_1 and I_2 flow in the opposite directions, therefor:

$$I_3 = -(-I_1) - I_2$$
$$\text{if} \quad |I_1| = |I_2|$$
$$\text{then} \quad I_3 = 0$$

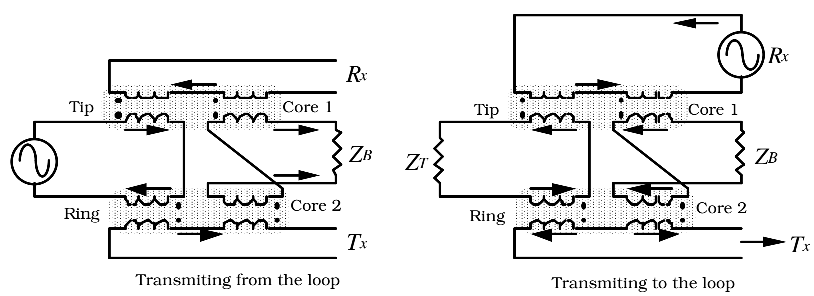
This last requirement can be satisfied by adjusting the impedances Z_1 - Z_4 to make the currents equal. From this we observe that signals injected into any port emerge only at adjacent ports but not at the opposite one.

In a properly balanced single core hybrid the typical throughput or insertion loss is about 3.5 dB and the THL† is about 25 dB.



Double Core Hybrid

When properly balanced, a 2-core network can achieve a THL of 50 dB while the insertion loss remains at about 3.5 dB. It has better performance than the single core device, but is bulkier and more expensive.



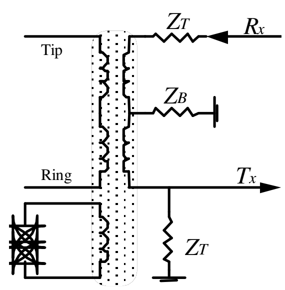
† Trans Hybrid Loss

_____ Notes _____

Flux Cancellation

In order to keep the transformer size as small as possible, high permeability materials such as ferrite, are used for the core. This allows the transformer to have a very high primary inductance. The inductance must be sufficiently large that its reactance in the voice band can largely be ignored, when compared to the impedance reflected across the transformer.

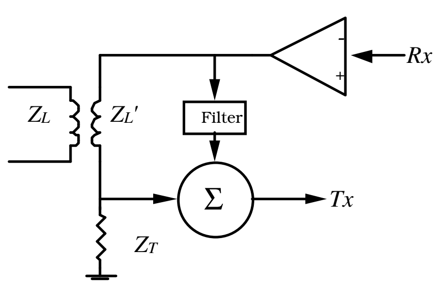
Since DC loop current flows through the transformer when the customer is off-hook, a static magnetic field is built up. This can cause the transformer core to saturate and act like a short circuit. To counteract this phenomenon, a separate winding is used to create a cancellation flux.



An exact duplicate of the DC loop current is created and fed into the flux cancellation winding. However, since the primary [or loop] current can be as high as 100 ma, this results in an enormous waste of power. To reduce this wastage, the flux cancellation winding has 4 to 5 times as many turns as the primary winding, thus requiring only 1/4 to 1/5 of the current necessary to provide cancellation since the flux is equal to the product of the current and number of turns.

Filter Hybrid

This technique may not use a transformer to couple the circuit to the line. Trans-hybrid cancellation occurs by generating in the filter the exact inverse of the signal going through the transformer from the *Rx* to the *Tx* port and then adding them. Thus, none of the *Rx* signal can enter the *Tx* port. The signals coming from the loop however, are not canceled since they pass through the transformer, but not the filter.



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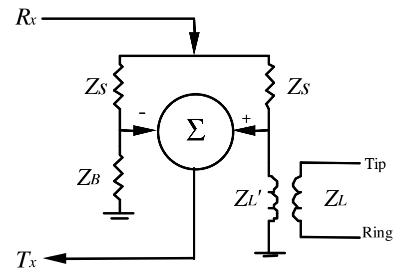
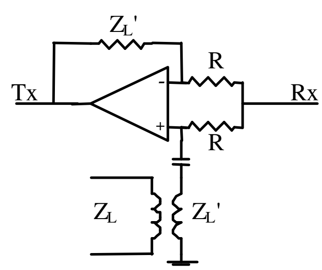
Impedance Hybrid



An impedance Z_B equal to the reflected line impedance Z_L'' is created and used to balance the hybrid. The balance impedance may be a simple RC network or it may be a more complex circuit implemented with opamp gyrators.

Miscellaneous Hybrids

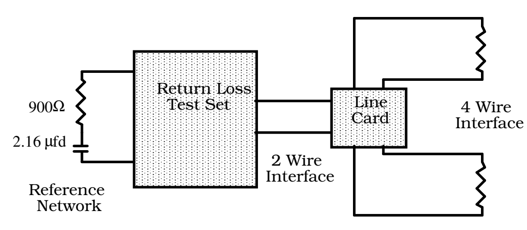
The following hybrid does not have very good performance, but it is adequate for FSK modems, where bandpass filters are used to separate transmit and receive signals. Balance is achieved by making the feedback element a duplicate of the reflected line impedance.



Balancing Networks

All telecom equipment is tested and characterized against standard impedance terminations. These impedances are based on line surveys and are approximate equivalent circuit representations of the outside cabling plant. For this reason, these networks vary from country to country.

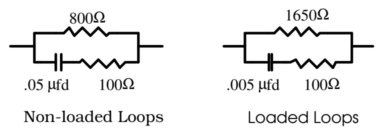
In North America, IRL† is measured against:



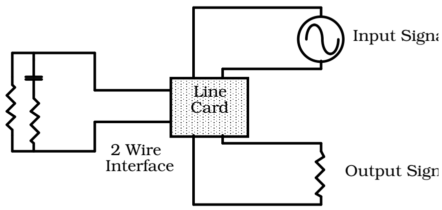
† Input Return Loss

______ Notes ______

Trans-hybrid loss is measured against the following reference networks:



Loaded loops have inductors place in series with the line to improve the passband frequency response. These coils are used only on voice grade rural loops.



Echo Cancellation


[Echo Cancellation by Tellabs]

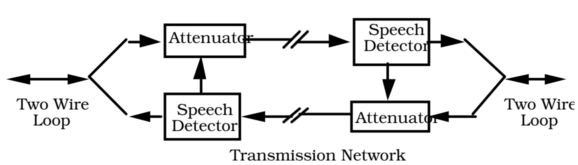
Electrical signals traveling down a transmission line are reflected back to the source when they encounter a change in impedance. This can result in unacceptable distortion. If a voice signal is returned to the talker (echoed) after a 30 mSec transmission delay, the voice path will seem hollow, and the talker gets the impression of speaking in a barrel.

In the extreme case, these echoes can return in phase, reinforce, and the lines oscillate or sing. This phenomenon is often observed in hands-free telephone sets when the speaker and microphone are in close proximity and the possibility of acoustic feedback is quite high.

On satellite circuits, the round trip echo delay is in the order of .5 seconds. If the echo is loud enough it becomes almost impossible to speak, since one is constantly interrupting oneself.

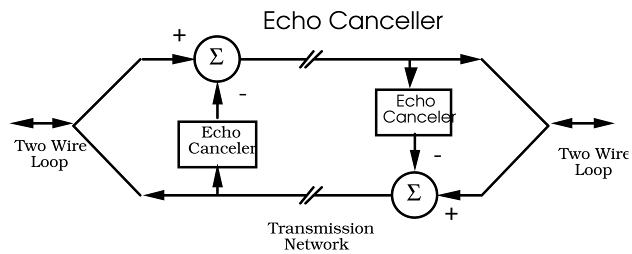
One way to eliminate the echo is to break the return echo path. An echo eliminator determines which person is speaking or which one is the loudest, and temporarily disables the other path. Conversation can be somewhat jerky, unless both parties recognize what is happening. In a poorly implemented system, syllables can be clipped and no amount of telephone courtesy can compensate.

Echo Suppressor

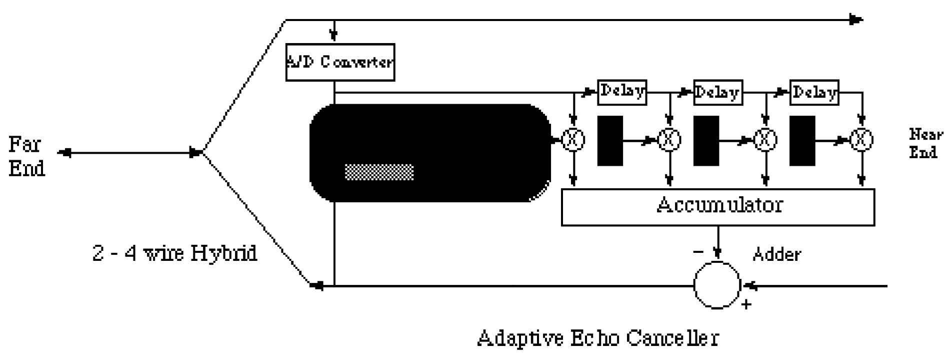


A less drastic approach is to provide a high loss in the return path. This compromise allows the far end talker to be slightly heard, but at the expense of letting some echo pass. This approach is often used on hands free sets as well.

Today, digital echo canceling is used. These circuits allow full duplex speech on long haul links. The processors in the echo canceller recreate the anticipated echo, and then subtract it from the incoming signal. These circuits are generally adaptive, and able to readjust themselves to meet a wide range of line conditions. They are also relatively expensive, and up till recently have been used exclusively on satellite links.



With some of the new telephony services such as ISDN, adaptive echo cancellation in two wire loops is essential.



2.4.2.8 Testing

In order to maintain a high degree of service, the equipment must be capable of detecting and repairing faults before the customer is even aware that there may be a problem. As a result, a separate test buss and access relay is provided on a line interface. Tests may be performed in a bridged mode or with the loop and line card disconnected from each other.

Testing can be done in three basic directions:

- From the line interface looking out towards the subscriber loop
- From the loop connection looking into the line card
- From the central office side of the line card

It is standard procedure that all line interface circuits in a modern telephone exchange be routinely tested. These tests are generally automated and are conducted late at night when there is little chance that the customer will request service, thus interrupting the test. Some of the tests that may be scheduled include:

- Transmit and receive levels
- Transmit and receive frequency response
- Insertion loss
- Trans-hybrid loss
- Quantizing distortion
- Aliasing distortion

Some other tests that may be performed when commissioning a line or when a complaint is lodged, include:

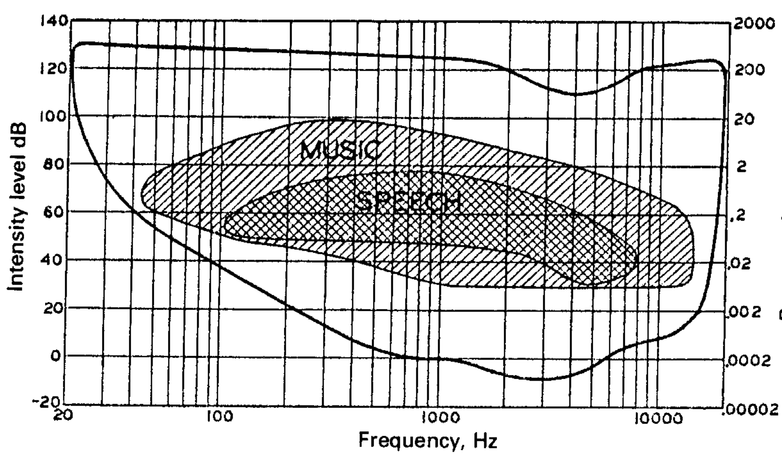
- Impulse noise test
- C-message noise
- Longitudinal balance

2.4.2.9 Voice Quality

Frequency Response

Many studies have been conducted to determine the required passband for recognizable voice. The frequency and energy content must be measured in order to determine the appropriate signals to send through the network.

Frequency and Intensity Ranges for Music and Voice[5]



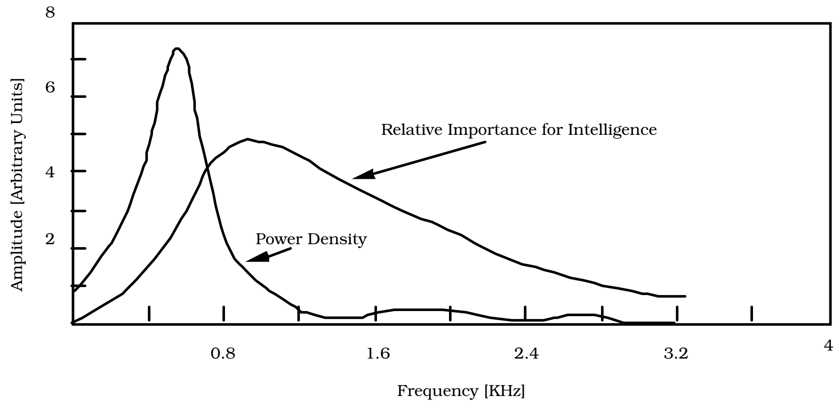
Commercial broadcast radio systems transmit music and voice. Since musical instruments produce a wider range of frequencies than voice, high bandwidth is required. By way of contrast, the power of the human voice is concentrated in a relatively narrow band.

5 *Bell Laboratories Record*, June, 1934

_____ Notes _____

Furthermore, some frequencies are more important than others in determining intelligibility. Ultimately it is the voice power density and intelligibility, which determine the minimum bandwidth requirement. It has been subjectively determined that a frequency channel ranging from 300 Hz to 3400 Hz has sufficient bandwidth to allow a person's voice to be both understood and recognized.

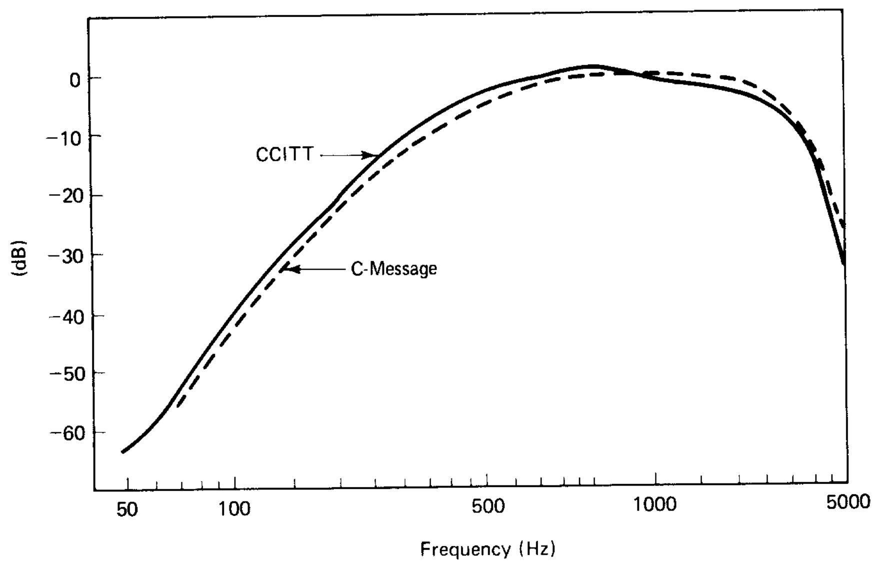
Speech Power and Intelligibility[6]



Several methods have been developed to determine whether the signal or noise levels in any particular place are acceptable. The requirements for music or data circuits are much more stringent than for voice.

6 Based on. *Digital, Analog, and Data Communication*, William Sinnema, Figure 1-5

C-Message Curve[7]



The standard telephone loop itself does not limit the voice passband. Rather, the BORSCHT interface connected to the loop does.

Since the highest frequency passed is about 3.4 KHz, a great deal of ingenuity must be used to pass data at 56 Kbps. Note that these are well above the Nyquest rate but considerably below the Shannon-Hartley limit.

All modern telephone systems today employ codecs at the BORSCHT interface to digitize the incoming analog signals. As mentioned in section 3, the μ-255 standard is used in North America. By international agreement, all voice codecs use an 8 KHz sampling rate. Since each sample is 8-bits long, the analog voice signal is translated into a 64 Kbps binary steam. This bit rate forms the basis of almost all multiplexing schemes.

By bypassing the codec, it is possible to send 64 Kbps customer data through the telephone system. However, because of old style signaling schemes still in use, this is more commonly limited to 56 Kbps.

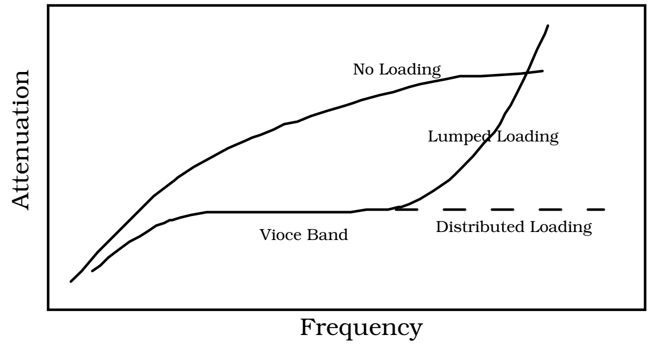
The wire gauge and the loop length have a significant impact on the attenuation distortion verses frequency curve. In order to flatten out the voice passband, loading coils are sometimes placed in the loop. These attenuate the high frequencies, and must be removed for data transitions.

Equalizers are the preferred solution today.

7 *Digital, Analog, and Data Communication*, William Sinnema, Figure 1-6

______ Notes ______

Loaded and Non-loaded Response[8]



Common Loading Types[9]

Loading Designation	Spacing in feet	Inductance in mH
B88	3000	88
B135	3000	135
H88	6000	88
H135	6000	135
H175	6000	175
M88	8000	88

2.4.2.10 Service Adaptive Access

The development of DSP[†] ICs and integrated high voltage bipolar transistors has revolutionized BORSCHT designs. These two things have led to the development of SLIC[†] chips for the central office environment. These chips are implemented in BiCMOS technology, which combines high voltage bipolar transistors on the same substrate as CMOS devices.

The next generation of SAA interfaces is apparently going to be implemented in a 0.8-micron process[10] known as BATMOS[†].

In the past, SLICs were developed for PBX environments where the loop environment is well defined, and there is little risk of exposure to lightning strikes. The development of high voltage bipolar technology has not only eliminated damage due to lighting but has also allowed the ringing function to be placed on the line card.

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8 Based on Figure 2-16, *Digital, Analog, and Data Communication*, William Sinnema,

9 *Digital, Analog, and Data Communication*, William Sinnema, Table 2-3

† Digital Signal Processing

† Subscriber Line Interface Circuit

10 Northern world-class IC technology keeps system costs in line, ep&t, Nov/Dec 1993, page 51

† Bipolar Analog Telecom Metal Oxide Semiconductor

DSP chips allow the line interface to adapt the coding, hybrid, and testing functions to meet a wide range of applications. The circuits developed by BNR and manufactured by NT are discusses further in the chapter dealing with PSTN Examples.

CLASS

 Enhanced Services Messaging by Comverse

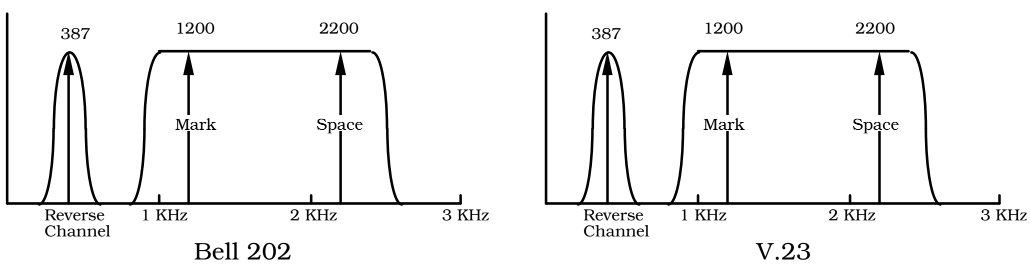
Telephone companies offer two broad categories of special services. One set does not require any modification of the subscriber set, while the other does.

Services such as call waiting, three party calling, call forwarding, redial, and so on, do not require modifications to the phone. These functions are all software driven and can be invoked by generating a hook flash at the appropriate moment or by dialing special access codes. Most telcos however, prefer to offer these types of services in conjunction with DTMF signaling.

With the development of digital over voice transmission and liquid crystal displays, a wide range of new telephone sets and services can be offered. Among the new special services is caller identification.

Caller ID[11]

This class of services is supported by utilizing continuous phase BFSK modulation conforming to the Bell 202 or CCITT V.23 standards.



Data messages are sent during the silent interval between the first and second ring. The message structure that is transmitted is of the form:

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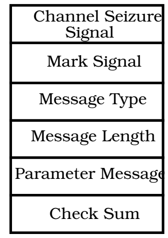
11 Caller ID chip makes new telecom services feasible, ep&t, Nov/Dec 1993, page 50

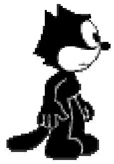
_____ Notes _____



Each one of these data fields has a unique purpose:

- Channel seizure signal: This is used for on-hook data only. It consists of a series of 300 alternating 1s and 0s, starting with 0 and ending with 1.
- Mark signal: This is 80±10 bits of continuous high.
- Message type: An 8-bit byte.
- Message length: A 1-byte word specifying the length of the subsequent message.
- Parameter message: Each of *N* parameters are broken down into three sub-fields: parameter type, length, and data. Some of the data that is included may be date, time, incoming call number, reason for absence of call number.
- Check sum: This is a 1-byte 2's complement sum of fields 3, 4, and 5, mod 256.

Assignment Questions



SOLUTIONS

QUICK QUIZ

1. The [first three, last four] digits in a telephone number are known as the exchange number.
2. [DC, AC] loop current is monitored by the far-end-office during ringing.
3. If a subscriber stays on the line when a busy tone is encountered, the connection will be completed if the calling party simply waits until the line is free. [True, False]
4. Telephone cables are stored in a cable vault. [True, False]
5. The nominal battery voltage in a telephone exchange is - [12, 24, 48] volts dc.
6. Most of the energy in the human voice is concentrated at about [600, 800, 1000] Hz
7. The frequencies, which govern the intelligibility of the human voice, are centered at [1, 2, 3] KHz.
8. Voice signals are digitized at [8, 12, 64] K samples per second.
9. The digitized voice bit rate is [9.6, 56, 64] Kbps.

10 The nominal ringing voltage in Canada is [68, 86, 110] volts rms.

11. A flux cancellation current of [4, 10, 40, 160] ma is required to compensate for a 40 ma loop if the hybrid transformer has a 4:1 winding.
12. An echo canceller attenuates the return signal path. [True, False]

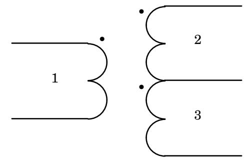
ANALYTICAL PROBLEMS

1. What is the traffic intensity in Erlangs for a 120 K line office, if the average calling rate during the busy hour is 2.1 calls per line, and the average call holding time is 6 minutes and 12 seconds?

Notes

2. Given the basic transformer equations:

$$i_1 n_1 + i_2 n_2 + i_3 n_3 + \ldots = 0 \qquad \frac{v_1}{n_1} = \frac{v_2}{n_2} = \frac{v_3}{n_3} = \ldots$$



a) Show that the figure illustrated above can act as a 2 to 4 wire hybrid.

b) Show how this device can be used in both line interface and repeater applications.

3. Each hybrid in a two-wire repeater has a THL of 37 dB, and insertion loss of 3.5 dB. If the repeater requires a 4 dB singing margin, determine the maximum theoretical repeater gain.

4. Given that the total number of crosspoints in a non-blocking network is

$$2N(2n-1)+(2n-1)\left(\frac{N}{n}\right)^2$$

Find the optimum, value for *n*, and the total number of crosspoints required, if 1000 subscribers are to be connected.

Composition Questions

1. What is the purpose of dial tone?
2. What are the advantages of CAS and CCIS interoffice signaling?
3. What is the difference between a line and a trunk?
4. Why does the standard telephone exchange need to provide BORSCHT?
5. Discuss the differences between North American and European codecs.
6. Under what circumstances can poor THL be tolerated?
7. What is the difference between an echo suppresser and an echo canceller?
8. Where do the telephony terms tip and ring originate?
9. Define blocking.
10. Why are multiple stage space (or time) switches more practical than a single stage switch?

Research Questions

1. Find the specification sheets for an integrated BORSCHT chip such as the *Motorola MC3419* and list some of its basic operating parameters.

Notes

2. Using the linked documents in this chapter, list the basic call setup procedure on a CC7 network.

_____ Notes _____

For Further Research

Bigelow, Stephen J., Understanding Telephone Electronics, Sams, 1991

Freeman, A. H., "Hybrid Transformers Part 1", TJA, vol. 27, no. 3 (1977)

Freeman ,A. H., "Hybrid Transformers Part 2"; TJA, vol. 28, no. 2 (1978)

Friedheim, H. O., "On Hybrid Transformers", ATE Journal vol. 14, no. 3 (July 1958)

Nishizuka, Honma, and Sato, "Analysis on Compensated Three Winding Hybrid Transformer;" Abstracts, The Transactions of the IECE of Japan, vol. E 62, no. 5 (May 1979)

Sartori, Eugine F., "Hybrid Transformers", IEEE Transactions on Parts, Materials, & Packaging, (September 1968)

General Telecom Links

http://china.si.umich.edu/telecom/internet-telephony.html

Standards

http://www.cmpcmm.com/cc/standards.html

The PSTN

http://www.cmpcmm.com/cc/standards.html

http://www.lidoorg.com/pstn.htm

http://www.phonezone.com/tutorial/index.htm#phones

Telephone Technology

http://www.mantis.co.za/elec/telephone.html#how

http://www.cybercom.com/~chuck/telhist.html

http://www.electronickits.com/kit/complete/tele/ck600.htm

http://www.navyrelics.com/tribute/index.htm

Telecom Libraries or Links

http://www.cybercomm.net/~chuck/phones.html

http://www.onepassinfo.com/opref2.html

http://www.wcom.com/library.html

http://www.analysys.co.uk/vlib/

http://www.ee.umanitoba.ca/~blight/telecom.html

http://china.si.umich.edu/telecom/telecom-info.html

Telecom Magazines

http://www.castle.net/~kobrien/trade.html

Telecom Cabling

http://www.bicsi.org/

AIN

http://www.ddx.com/ain.shtml

Telephony Terminology

http://www.its.bldrdoc.gov/fs-1037/fs-1037c.htm