CHAPTER CONTENTS

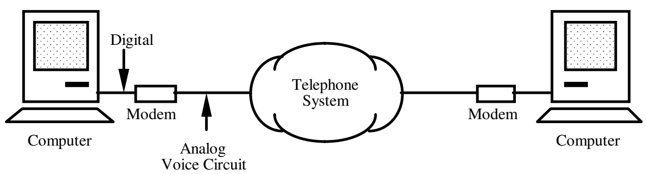
4.0 Telephone Circuits

Objectives

This section will endeavor to:

- Examine the nature of analog telephone loops
- Review telephony transmission hierarchy

The telephone system forms the largest communications network in the world. This facility was originally designed to handle analog audio signals but with the advent of computer technology, it became necessary to carry digital data. One way to do this is to make digital signals resemble voice signals; this is a principle function of a **MODEM**. Consequently, the modem is designed to match the characteristics of the analog telephone environment.



4.1 Voice Circuits

[TR_60 Unbundled Voice Grade Telephony Loops]

[DMS Line Card]

http://www.legerity.com/tutorial1.html

The standard telephone loop consists of a twisted pair of wires with a telephone on one end, and a BORSCHT circuit on the other. The **BORSCHT** circuit is in the telephone central office and provides a number of useful services:

B Battery feed

O Over voltage protection

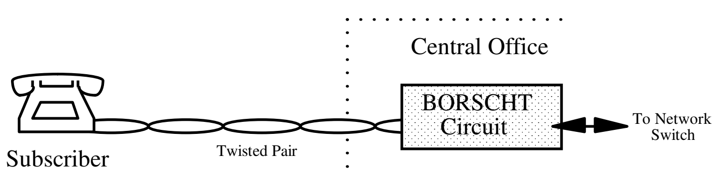
R Ringing

S Supervision & Signaling

C Coding

H Hybrid

T Test



Telephone cables vary from 19 to 26 AWG, and may contain 6 to 2700 pairs of wires. A telephone line is a balanced two-wire transmission line. The impedance from each wire to ground is generally matched to within 60 dB. As a result, any induced noise is common mode or longitudinal in nature and can be easily removed.

A massive -48 volt battery in the telephone exchange powers the telephone. This voltage is applied to the **RING** lead while the **TIP** wire provides the return path. A negative voltage generates less galvanic corrosion than a positive voltage.

4.1.1 Passband

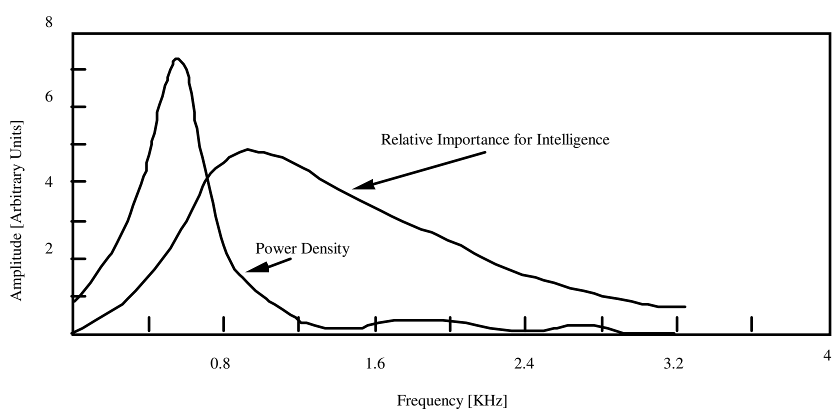
The audio bandwidth used in any system is determined by the application.

System	Approximate Audio Bandwidth [KHz]
FM Radio	15
AM Radio	5
Telephone	3.4

Commercial broadcast radio systems require relatively high bandwidth since musical instruments produce a wide range of frequencies. On the other hand, the power in the human voice is concentrated in a relatively narrow frequency range centered at about 500 Hz. Furthermore, some frequencies are more important than others in determining speech intelligibility.

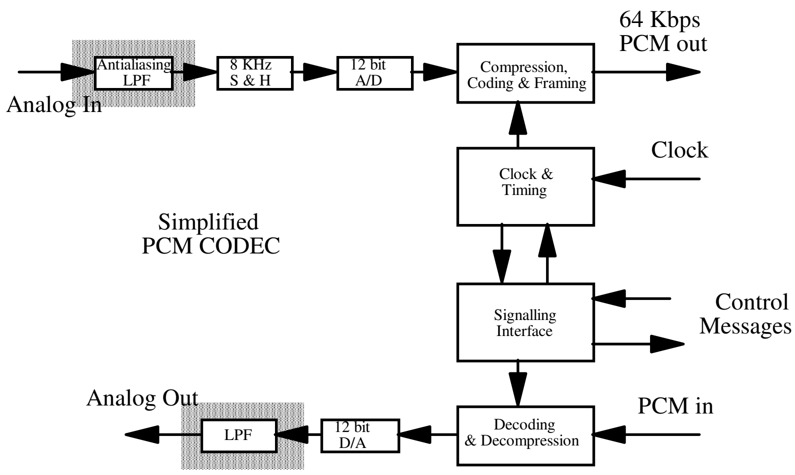
Ultimately it is the two basic characteristics voice power density and intelligibility, which determine the minimum bandwidth requirement. It has been subjectively determined that a frequency channel ranging from 300 Hz to 3400 Hz has sufficient bandwidth to allow a person's voice to be recognized.

Speech Power and Intelligibility[1]



The anti-aliasing filters in the codec (not the telephone line) limit the channel passband.

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



[TI - PCM Codec Filter Combo]

[AMD - SLIC & Longitudinal Balance]

[AMD - Ringing SLIC]

Since the highest frequency passed is about 3.4 KHz, a great deal of ingenuity is required to pass data at 4.8, 9.6 Kbps or even higher. Note that these are well above the Nyquest rate but considerably below the Shannon-Hartly limit.

All modern telephone systems today employ codecs in the BORSCHT interface to digitize the incoming analog signals. It is ironic that although the telephone system has been updated to digital technology, the telephone set and loop has remained analog.

By international agreement, all voice codecs use an 8 KHz sampling rate. Since each transmitted sample is 8 bits long, the analog voice signal is encoded into a 64 Kbps binary steam. This rate determines the basic channel data rate of most other digital communications systems.

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, digital data rates are often limited to 56 Kbps.

4.1.2 Signal & Noise Levels

4.1.2.1 Reference Frequency

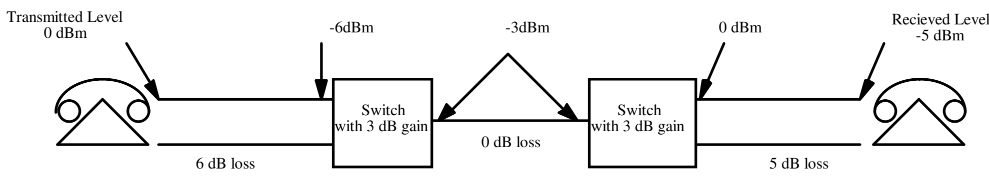
In North America, the standard test or reference frequency is 1004 Hz. 1 KHz is not suitable since it is a sub-harmonic of the 8 KHz sampling used in the D/A conversion process.

4.1.2.2 TLP†

All signal handling systems require a reference level point from which to make measurements. In a telephone network, this level is defined as 0 dBm and is located at the **MDF**†. All measurements made with respect to this reference are designated as dBm0.

A signal looses power as it passes through a network, and must be boosted periodically. Consequently, both signal and noise levels vary along the connection path.

The total attenuation on a voice connection should not exceed 16 dB.



Level Changes Through a Connection

4.1.2.3 Noise Reference

The most common noise found in any communications channel is random white noise with a Gaussian [normal] distribution. It is caused by thermal agitation. The standard reference level used for making noise measurements is 1 pW or -90 dBm.

A signal level of -90 dBm is considered quiet. Signals measured with respect to it are designated as dBrn. A 0 dBrn signal has a level of -90 dBm.

If a **C-message** filter is used in making a measurement, the signal designation is dBrnc. If the signal is also referenced to 0 TLP, the designation is dBrnc0.

4.1.2.4 C-Message Filtering

Since some noises are more disturbing than others, it is not necessary to treat all noise equally. For this reason, the C-message filter has been developed.

The C-message filter is a bandpass filter with a 0 dB loss at 1 KHz, and a 5 dB loss at about 450 Hz and 3 KHz. It will reduce a 300 Hz to 3400 Hz white noise source by about 1.5 dB. Power measurements made with this filter are designated in dBmC. The objective on a PSTN voice channel is to achieve 28 dBrnc for connections of 60 miles or less, and 34 dBrnc for circuits that are 1000 miles long.

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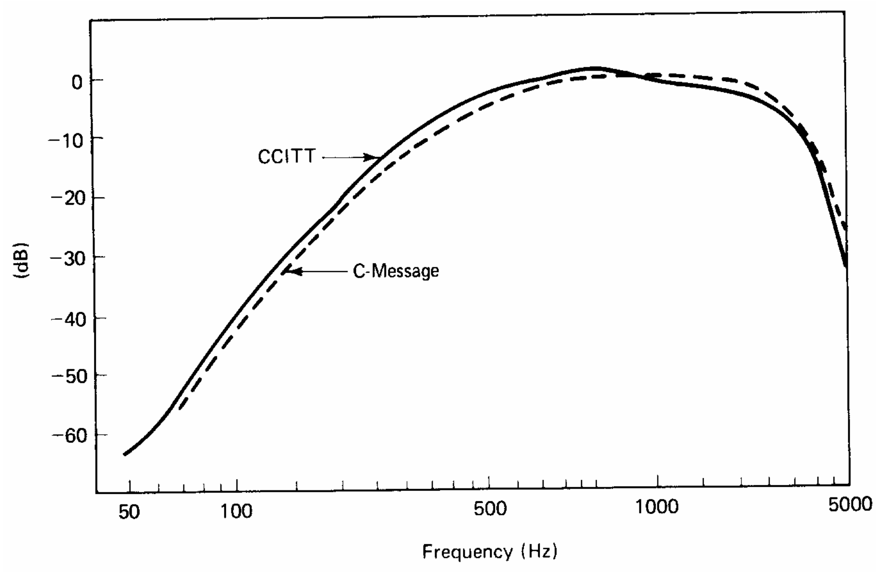
† Transmission Level Point

† Main Distribution Frame

C-Message Curve[2]

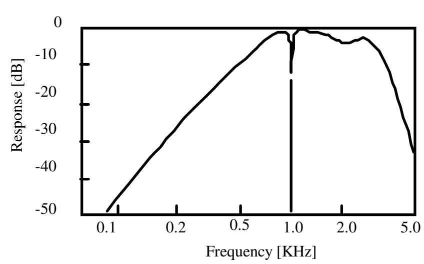


In Europe a slightly different filter known as a **psophometric** or CCITT filter is used. It has a reference frequency of 800 Hz. and will reduce a 300 Hz to 3400 Hz white noise source by about 2.5 dB. Power measurements made with this filter are designated in dBmp.

Noise measurements made on idle voice channels are of some value, but it is also beneficial to know the noise level when signals are present. These type of noise measurements take into account things such as quantization and aliasing noise which is not present if the codec is not being exercised, and harmonic distortion which may occur when amplifiers are operated at high levels.

A test tone of about 1 KHz is injected into the line, thus forcing codecs and amplifiers to operate over a large amplitude range. At the receiving end a notch filter eliminates the test tone and the remaining noise is measured.

Notched C-Message Curve[3]



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

3 Based on *Data Transmission* (2nd ed.) Tugal & Tugal, Figure 2.25

4.1.2.5 Impulse Noise

Impulse noise is most often associated with electromechanical exchanges but may also occur in electronic systems. Sharp noise spikes are somewhat annoying to listeners; consequently, pulse telephone sets have varistors to help reduce switching transients created by dialing. Impulse noise is of more serious concern to data customers, since it may contribute to higher error rates.

Signal & Noise Level Examples

Example 1

At a -22 dBm TLP, a technician measures a C-message noise level of -72 dBm and a test tone level of -25 dBm. Determine:

a) Tone signal power relative to TLP [dBmO]

b) C-message noise relative to reference noise [dBrnc]

c) C-message noise relative to reference noise at 0 TLP [dBrnc0]

d) Overall signal to noise ratio [dB]

Solution:

If we make a sketch of the various signal levels, their relationships becomes clear:

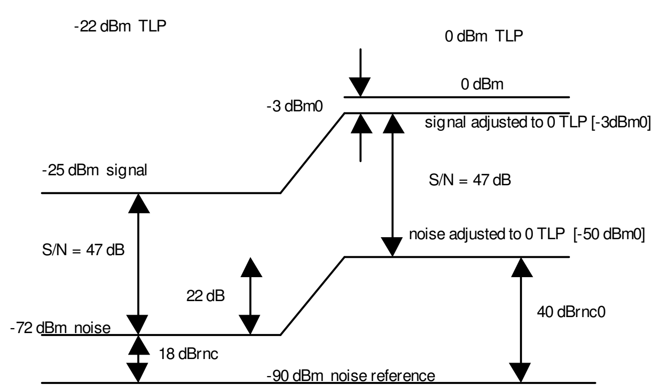
a) -3 dBmO

b) 18 dBrnc

c) 40 dBrnc0

d) 47 dB

Signal Level Diagram



Example 2

A noise level of 43 dBrnc0 is measured at a -20 dBm TLP and a test tone is used to confirm that the S/N is 31 dB. Find:

a) Signal relative to 0 TLP [dBmO]

b) C-message noise relative to reference noise [dBrnc]

c) Actual power of the signaling tone [dBm]

d) Actual C-message noise power [dBm]

Solution:

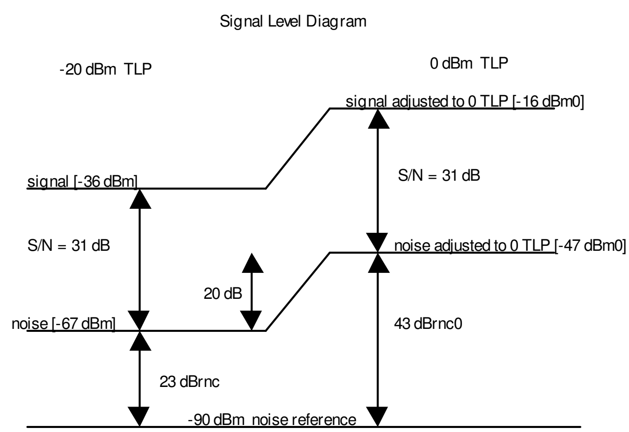
Make a sketch of the various signals and their relationships become almost clear:

a) -16 dBmO – The noise power level relative to TLP is - 90 dBm +43 dBmO = -47 dBmO. Therefor the signal level relative to TLP must be -47 dBmO + 31 dB = - 16 dBmO

b) 23 dBrnc – The C-message noise power relative to the reference noise and corrected to TLP is - 43 dBrnc0, and since the TLP is - 20 dB, the noise relative to the reference noise level is -43 dBmO - 20 dB = - 23 dBrnc. On the other hand, alternately, since the noise power relative to TLP [but not with respect to the noise reference] is - 47 dBm [see part a) above], then the actual noise power must be - 47 dBm -20 dBm = -67 dBm. Consequently, this noise relative to the reference noise is 90 dBm - 67 dBm = 23 dBrnc

c) -36 dBm – Since the actual noise power level is - 67 dBm [see part b) above] and the S/N = 31 dB, the actual signal power must be - 67 dBm +31 dB = - 36 dBm

d) - 67 dBm – [see part b) above]



4.1.3 Distortion

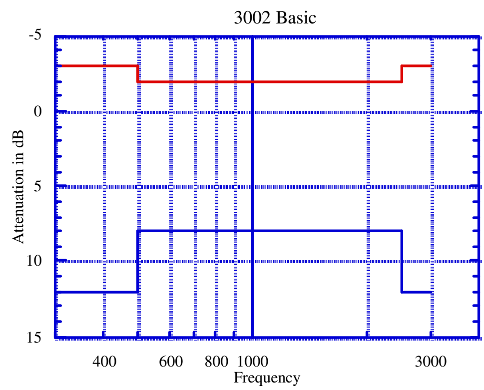
Unlike noise, distortion can be largely compensated for since it is generally deterministic.

4.1.3.1 Attenuation Distortion

A telephone set injects approximately 1 mW of voice power into the loop. This establishes the maximum size of data signals that can be injected into the loop.

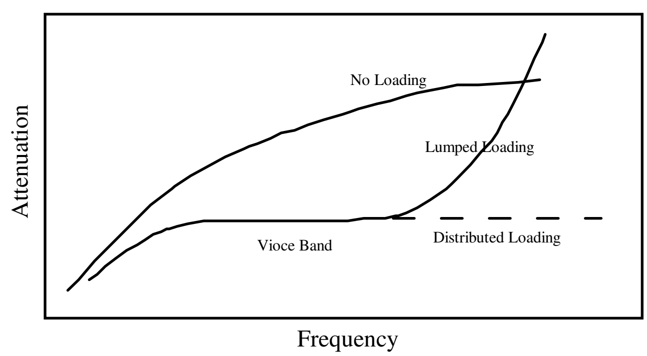
Modems with constant amplitude tones are limited to -10 dBm0 signal levels while those using amplitude modulation are restricted to -6 dBmO.

Once the signal is in the system, its path and environment can be very tightly controlled, but the outside plant is more difficult to regulate. The particular cable loss characteristics determine whether a line suitable for voice, is good enough for data.



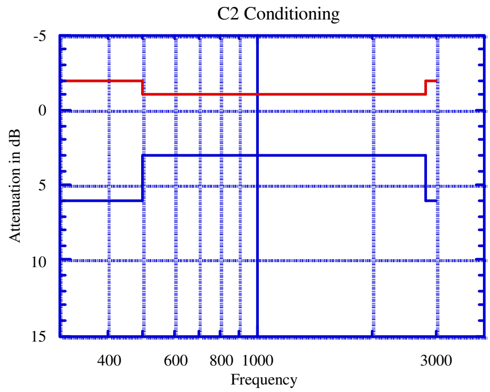
One method for improving audio attenuation distortion on long loops involves the use of loading coils. Although they significantly improve the voice path passband, they are unsuitable for data lines. The series inductance adversely affects the fast rise times associated with digital signals.

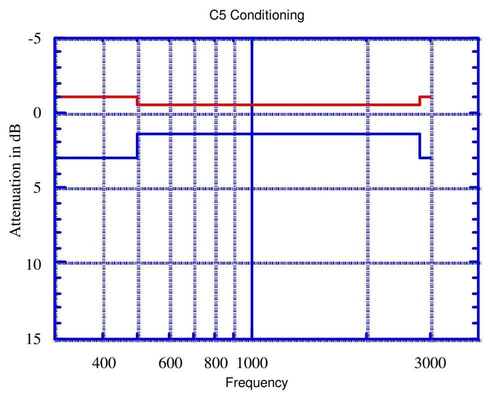
4.1.3.2 Loaded and Non-loaded Response[4]



Lines dedicated to transmitting data generally require tighter tolerance on attenuation distortion.

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





4.1.3.3 Harmonic Distortion

Since a telephone channel has a limited bandwidth, a relatively low frequency must be used as a reference source for this measurement. A 704 Hz tone is transmitted through the system. Any nonlinearities will generate harmonics at integer multiples of this rate namely, 1408 Hz and 2112 Hz.

A frequency selective voltmeter can be used to measure these individual components, and then express the result as a percentage of the original signal level. An alternate method is to measure the total noise with and without a 704 Hz notch filter in the circuit however, it would be necessary to eliminate the influence of all other noise sources.

4.1.3.4 Intermodulation Distortion

Nonlinearities can create signals other than pure harmonics. If two signals are present, sums and differences of all of the harmonics can be generated. This is a form of heterodyning.

To make this measurement, 860 and 1380 Hz tones are injected onto the line, and noise measurements at 520, 1900 and 2240 Hz are made. The combined noise of these spurious signals is then compared to the originating tones.

4.1.3.5 Delay Distortion

The propagation velocity of electrical signals down a transmission line is somewhat dependent on frequency. This phenomenon causes a frequency dependent phase shift.

An absolute phase shift can be measured by performing a far-end loop around and displaying the input and output signals as a Lissajous pattern. The phase shift at any given frequency is then measured by the tilt of the resulting ellipse. From this delay can be determined since it is the first derivative of phase.

$$\text{Delay} = \frac{\Delta\theta}{\Delta f} \times \frac{1}{360} \times 10^6 \mu s = \frac{\Delta\theta}{\Delta f} \times 2778 \mu s$$

A relative phase shift is much easier to measure than the absolute propagation velocity. This can be done by amplitude modulating a carrier signal, and then measuring the phase difference between the two sidebands. This parameter is known as the envelope or group delay, and can be calculated by using the same equation.

The standard technique modulates a carrier with a low frequency [25 or 831/3 Hz] tone. The modulation signal is removed at the far-end and used to modulate a second carrier back to the test set. The test set then compares the phase difference between the transmitted and received modulation signal.

4.2 Impedance Matching

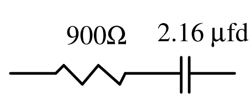
All telecom equipment is tested and characterized against standard impedance terminations. These impedances are based on line surveys and are approximate equivalent circuit representations of the outside cabling plant. For this reason, these networks vary from country to country.

Impedance matching is important for a number of related reasons:

- To provide maximum signal transfer
- Minimize echo
- Prevent oscillations

4.2.1 Input Return Loss

The input impedance presented to the telephone network must determine how much of the signal will be reflected back by the near-end connection. This condition is called near-end echo. In North America, input return loss is compared to a reference impedance network [Z_o] of:



while the far-end is terminated in 600 Ω. The return loss is determined by:

$$\text{Return Loss} = -20\log\left|\frac{Z_{in} - Z_o}{Z_{in} + Z_o}\right| \text{dB}$$

The minimum return loss should be 26 dB in the frequency range of 500 - 3400 Hz. There are other types return losses, but they are primarily of concern to the telephone companies themselves.

One of these is transhybrid loss. Its primary effect is on far-end echo and singing.

4.2.2 Far-End Echo and Singing

Echo occurs when a signal returns to the originator. In extreme cases, oscillations or singing may occur. Echo is often a result of transhybrid impedance mismatches at the BORSCHT interface.

Both the magnitude and delay of the returning signal contribute to the annoyance. To minimize this, a net loss must always be maintained on any link. The loss on a local loop is set to between 2 and 4 dB. Round trip loss on toll trunks is set to:

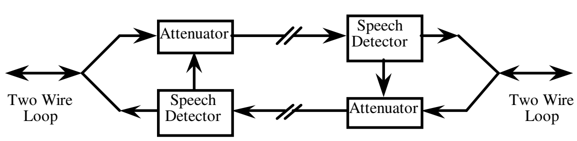
$$VNL = 0.1 \times \frac{2l}{v} + 0.4 \ \text{dB}$$

$$l = \text{circuit length in Km}$$

$$v = \text{velocity of propagation in Km/mSec}$$

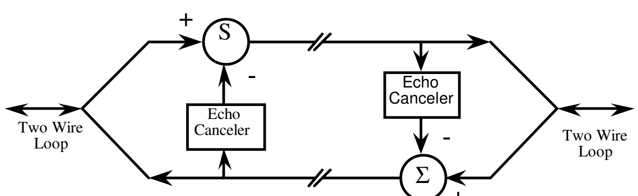
Circuits with a round trip delay in excess of 45 mSec require an unacceptably high level of attenuation for echo control. For this reason, a 35 dB attenuator is placed in the return circuit if the forward circuit power level exceeds a specified threshold. These suppressers should have a response time of 2 to 5 mSec.

Echo Suppressor



A superior but more complex method involves the use of echo cancellers. It does this by exciting the echo path and then creating an attenuated and delayed copy of the return signal. This can be subtracted from the incoming signal when the path is established.

Echo Canceler



Until recently, these circuits were extremely expensive, physically large and were found only on satellite links. However, with the advent of DSP and VLSI, these devices are being found everywhere echo is a concern.

Echo Cancellation Tutorial by Coherent

http://lcavwww.epfl.ch/~prandoni/dsp/echo/echo.html

http://www.signalworks.com/

http://www-nt.e-technik.uni-erlangen.de/audio/research.html

4.3 Digital Trunk Circuits

TDM is a convenient method for combining various digital signals. In the case of voice circuits, these signals are generated by codecs. Each 8-bit codec output is called a channel. Channels in turn are organized into frames.

The frames are comprised of channels interleaved at the bit, byte, or some other level. The resulting pattern may be transmitted directly, as in digital carrier systems, or passed through a modem to allow the data to pass over an analog network.

Digital data is generally organized into frames for transmission and individual users assigned a time slot, during which frames may be sent. If a user requires a higher data rate than that provided by a single channel, multiple time slots can be assigned.

Digital transmission schemes in North America and Europe have developed along two slightly different paths, leading to considerable incompatibility between the networks found on the two continents.[5]

BRA Comparison		
Characteristic	North America	Europe
Basic channel rate	64 Kbps	64 Kbps
CODEC format	μ-Law	A-Law
Binary format	Folded binary	Sign magnitude
Channels per frame	24	30
Signaling	Bit robbing	Dedicated channel

5 *The Evolution of the Digital Loop Carrier,* IEEE Communications Magazine, March 1991

Digital Transmission Systems[6]

Designation	Bit Rate [Mbps]	Line Code	Media	Repeater Spacing
T1	1.544	AMI/B8ZS	Twisted pair	6 Kft
CEPT1	2.048	HDB3 (B4ZS)	Twisted pair	2 Km
T1C	3.152	Bipolar	Twisted pair	6 Kft
T148	2.37 ternary	4B3T	Twisted pair	6 Kft
9148A	3.152	1-D^2 duobinary	Twisted pair	6 Kft
T1D	3.152	1+D duobinary	Twisted pair	6 Kft
T1G	6.443	4-level	Twisted pair	6 Kft
T2	6.312	B6ZS	Low cap twisted pair	14.8 Kft
LD-4	274.176	B3ZS	Coax	1.9 Km
T4M	274.176	Polar	Coax binary NRZ	5.7 Kft

4.3.1 North American TDM Carriers

The various transmission bit rates are not integer multiples of the basic rate. This is because each multiplexing level requires additional framing and synchronization bits.

DS Hierarchy			
Designation	# Voice Channels	Bit Rate [Mbps]	Comments
DS[†]–0	1	.064	100% duty cycle, unipolar
DS–1	24	1.544	Media: 22 AWG cable Repeater spacing: 6000 ft Signal: bipolar RZ
DS–1C	48	3.152	Media: 22 AWG cable Repeater spacing: 6000 ft Signal: bipolar RZ
DS–2	96	6.312	Media: low capacitance 22 AWG Max repeater spacing: 14,800 ft Signal: bipolar RZ, B6ZS
DS–3	672	44.736	Media: microwave, fiber Signal: RZ, B3ZS
DS-3A	1344	91.04	
DS–4	4032	274.176	Media: coax, microwave Signal: bipolar NRZ

Fixed rate TDM networks do not require the flow control mechanisms found in packet networks and error control is performed on a channel basis.

To aid in identifying the beginning of a frame, a framing pulse or a control channel is inserted into the data stream. In spite of this, it is possible for the transmitter and receiver to get out of sync. In such a case, all channels are lost as the receiver enters a search mode.

6 *Digital Telephony* (2nd ed.), John Bellamy, Table 4.7

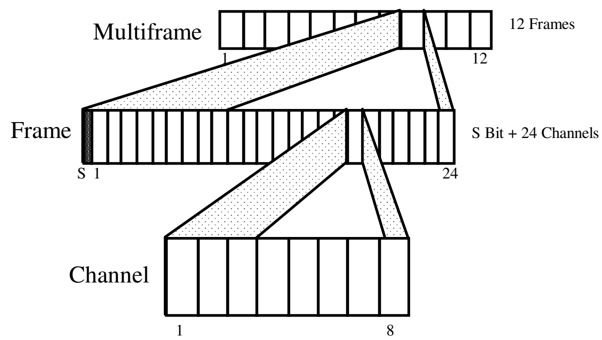
† Digital Signal type 0

Loss of lock can occur if noise disrupts the sequence, or if the transmitter and receiver clocks run at slightly different rates. It may be necessary to stuff the occasional bit into the data stream to resynchronize both ends of the link.

4.3.1.1 DS-1 24 Channel System

In North America, the basic digital channel format is known as DS-0. These are grouped into frames of 24 channels each.

A concatenation of 24 channels and a start bit is called a frame. Groups of 12 frames are called multiframes or superframes. These vary the start bit to aid in synchronizing the link and add signaling bits to pass control messages.



T-1 Multiplexing animated GIF

4.3.1.2 S Bit Synchronization

The S bit is used to identify the start of a DS-1 frame. There are 8 thousand S bits per second. They have an encoded pattern, to aid in locating channel position within the frame.

Frame	1	2	3	4	5	6	7	8	9	10	11	12
S Bit	1	0	0	0	1	1	0	1	1	1	0	0

This forms a regular pattern of 1 0 1 0 1 0 for the odd channels and 0 0 1 1 1 0 for the even channels.

Additional synchronization information is encoded in the DS-1 frame when it is used for digital data applications, so lock is more readily acquired and maintained.

For data customers, channel 24 is reserved as a special sync byte, and bit 8 of the other channels is used to indicate if the remaining 7 bits are user data or system control information. Under such conditions, the customer has an effective channel capacity of 56 Kbps.

To meet the needs of low speed customers, an additional bit can is robbed to support sub-rate multiplexer synchronization, leaving 6 x 8 Kbps = 48 Kbps available. Each DS-0 can be utilized as:

- 5 x 9.6 Kbps channels or
- 10 x 4.8 Kbps channels or
- 20 x 2.48 Kbps channels.

In the DS-2 format, 4 DS-1 links are interleaved, 12 bits at a time. An additional 136 Kbps is added for framing and control functions resulting in a total bit rate of 6.312 Mbps.

4.3.2 ZBTSI†

ZBTSI is used on DS–4 links. Four DS-1 frames are loaded into a register, and renumbered 1–96. If there are any empty slots [all zeros], the first framing bit is inverted and all blank slots are relocated to the front of the frame. Channel 1 is then loaded with a 7-bit number corresponding to the original position of the first empty slot. Bit 8 used to indicate whether the following channel contains user information or another address for an empty slot.

If there is a second vacancy, bit 8 in the previous channel is set, and the empty slot address is placed in channel 2. This process continues until all empty positions are filled.

The decoding process at the receiver is done in reverse. Borrowing 1 in 4 framing bits for this system is not enough to cause loss of synchronization and provides a 64 Kbps clear channel to the end-user.

4.3.3 Signaling

Signaling provides control and routing information. Two bits are taken from each channel in the multiframe. These are called the A and B bits.

The A bit is the least significant bit in each channel in frame 6, and the B bit is the least significant bit in each channel in frame 12. This provides a signaling rate of 666 $^2/_3$ bits per channel.

The quality of voice transmission is not noticeably affected when 2% of the signal is robbed for signaling. For data, it may be a different story. If the data is encoded in an analog format such as FSK or PSK, then robbing bits is of no consequence, but if the data is already in digital form, then robbing bits results in unacceptable error rates. It is for this reason that in North America, a 64 Kbps clear channel cannot readily be switched through the PSTN. This means that data customers are limited to 56 Kbps clear channels. This simple condition has a profound effect on the development of new services such as ISDN.

In most facilities, the A and B bits represent the status of the telephone hook switch, and correspond to the M lead on the E&M interface of the calling party.

A 1 or mark represents on-hook and 0 or a space represents off-hook. In a pulse dial phone system, the A and B bits follow the pulses. In this way, the system passes the routing information to the next office. When the called party answers, the returning A and B bits are set to 0 to signify that the call has been picked up. The system drops the connection when the bits in either direction are set to 1 for a prescribed length of time.

† Zero Byte Time Slot Interchange

4.3.4 ESF†

CCITT has modified the North American digital hierarchy for the deployment of ISDN, by means of recommendation G.704. ESF consists of 24 DS-0 channels in a frame, but groups them into a 24-frame multiframe instead of the usual 12 frame multiframe.

The S bit is renamed the F bit, but only 1/4 of them are used for synchronization. This is possible because of improvements in frame search techniques and allows more signaling states to be defined.

At the moment, bit robbing is still used for signaling over an ESF link, but with the advent of ISDN, it will not be permitted. Instead, channel 24 is used to support a D channel.

F Bit Function	Frame Position	Comments
Sync pattern	4, 8, 12, 16, 20, 24	Pattern: 001011
CRC check	2, 6, 10, 14, 18, 22	Is the remainder from modulo-2 division of all the bits in the previous frame by the binary polynomial x^6+x+1
Maintenance	All odd frames	Currently not specified, but will convey: maintenance, diagnostic, and status information

4.3.5 DS-1C

This signal contains two DS-1s plus a 64 Kbps framing and synchronization channel.

The two bipolar DS-1 signals are converted to a unipolar format, but one of them is inverted. They are then brought into alignment by bit stuffing and bit interleaving. A control bit is introduced at the beginning of every 52-bit group. The 26 control bit pattern defines a 1272 bit block, which is subdivided into several components:

4.3.5.1 Control Bit Pattern

M1	C11	F0	C12	C13	F1
M2	C21	F0	C22	C23	F1
M3	C11	F0	C12	C13	F1
M4	C21	F0	C22	C23	F1

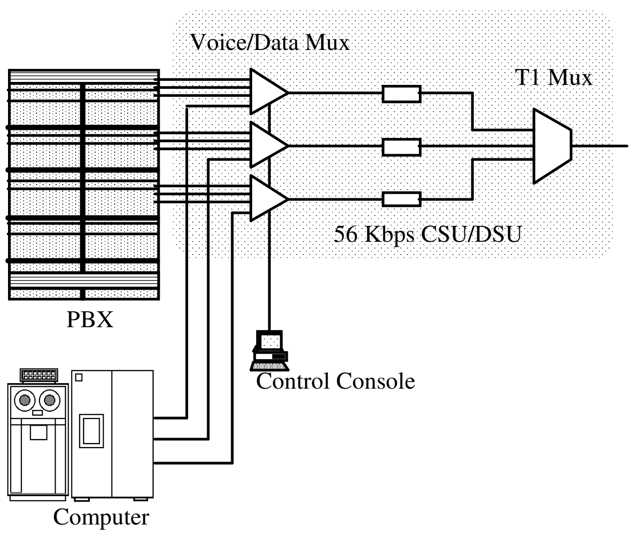
† Extended Super Frame

Bits	Function
M1, M2, M3	These bits are set to 011 and define the M frame
M4	This forms a signaling channel and is set to 0 if there is an alarm condition
F	This sequence has alternating 1s and 0s appearing at the beginning of every third 52 bit group
C	This bit identifies the presence of stuffed bits

4.3.5.2 Typical T1 CPE Application

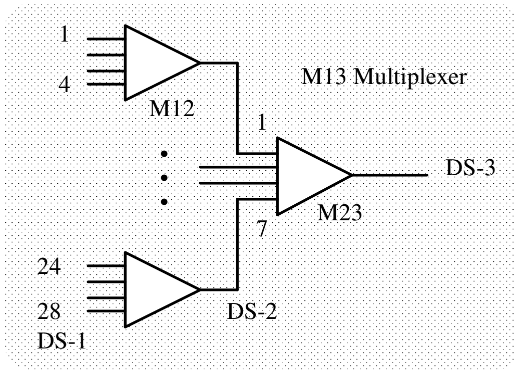
The large telecom carriers are not the only ones who deploy high-speed TDM facilities. In many cases, heavy users of voice or data services can reduce their transmission costs by concentrating their numerous low speed lines on to a high-speed facility.

There are a wide variety of T1 multiplexers available today. Some are relatively simple devices, while others allow for channel concatenation, thus supporting a wide range of data rates. The ability to support multiple DS-0s allows for easy facilitation of such protocols as the video teleconferencing standard, Px64.



4.3.5.2 Multiplexers

Multiplexing units are often designated by the generic term M*ab* where *a* is input DS level and *b* is the output DS level. Thus, an M13 multiplexer combines 28 DS–1s into a single DS–3 and an M23 multiplexer combines 7 DS–2s into a single DS–3.


1
4
M12
M13 Multiplexer
1
DS-3
M23
7
24
28
DS-2
DS-1

Review Questions

Quick Quiz

1. The reference noise level of -90 dBm corresponds to a power level of _____________ watts.
2. The noise level of 20 dBrn corresponds to a power level of _____________ watts.
3. What are the two main purposes of the signaling channels on a DS-1 link?
4. A DS-1 link has [12, 24, 48] voice channels.
5. The start bit in most communications systems is active [high, low].

Analytical Problems

1. At a -22 dB TLP in a communications network, it is found that the signal level is -25 dBm and the noise level is 13 dBrnc. What is the S/N ratio?

Composition Questions

1. Define C - message weighting.
2. What is the purpose of making notched noise measurements?
3. What is the speech bandwidth for a telephone and why is it band limited?
4. What is the noise reference level?
5. What are A and B bits, and how do they affect data customers?
6. How will ISDN impact on the ESF?
7. What is the basic limitation caused by the signaling method on a DS-1 link?
8. Why does the extended super frame require only 1/4 of the synchronization bits of the DS-1?
9. What is the principle cause of echo in telephony circuits?
10. How is delay distortion measured?

SystemView Models

1. Create an analog 8-channel PSTN voice channel TDMA mutilplexer model.
2. Modify the TDMA model in problem 1 for 64 Kbps digital channels.

For Further Research

http://www.intersil.com/

http://www.nortel.com/

http://www.lucent.com/

http://www.ssc.siemens.com/

http://www.itecinc.com/tele.html

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