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NEAR EAST TECHNICAL UNIVERSITY
ELECTRIC AND ELECTRONIC ENGINEERING DEPARTMENT

PULSE-CODE MODULATION TECHNIQUE FOR DIGITAL TELEPHONE

SUPERVISOR

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The authors also wishes to express his deep gratitude to his parents for their support throughout his life.

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INTRODUCTION

Pulse - code modulation (PCM) was developed in 1937 at the Paris Laboratories of American Telephone and Telegraph Company (AT&T) by Aled Reeves. Reeves conducted several succesful transmission experiments across the English channel using the modulation techniqe, including pulse-width modulation (PWM), pulse-amplitude modulation (PAM) and pulse-position modulation (PPM). At the time circuits involved was enormously complex & expensive. Although the significance of Reeve's experiments was acknowledged by Bell laboratories, it was no until the semiconductor industry evolved in the 1960s that PCM because more prevalent. Currently in the world PCM is the prefered method of transmission within the Public Swiched Telephone Network (PSTN).

PCM is a methode of digital transmission of an analog signal. The motives for using PCM in the telephone network can be summarized in the following 7 points:

❑ *Transmission quality almost independent of distance.* A characteristic of a digital signal is its immunity to interference. Digital signals can be regenerated at intermediate points along a transmission line without loss of quality. This is not the case with analogue signals where not only the signal but also the noise is amplified at the intermediate amplification points. In the digital case the repeaters have to make the simple decision as to whether an incoming pulse is a "one" or a "zero". After the decision has been taken a fresh pulse is transmitted. It is certain that a number of incoming pulses will be so distorted that they cannot be recognized correctly, but this failure rate can be made as low as required. It should however be observed, that the PCM-systems used in practice and specified by CCITT are not, from a transmission quality point of view better than the FDM-system specified by CCITT.

❑ *Time-division-multiplex.* The TDM principle allows an increase in capacity on cable pairs originally used for single telephone channels. It may be feasible to introduce PCM transmission on these pairs instead of laying new cables.

❑ *Economy for certain links.* In certain applications, especially in the junction network, PCM transmission has shown itself to be competitive with any other method of transmission. The length of the transmission links should be in the inttermediate region where normal voice frequency (VF) links tend to be too long and FDM links too short.

This optimal distance interval is very dependent on local factors such as subscriber density, topology of the country etc., and therefore varies widely. However, the figures in the graph can be taken as being typical for the first order PCM multiplex system.

❑ *Economy in combination with digital switching.* A high proportion of the cost for PCM-systems lies in the terminal cost. The introduction of digital switching will lsubstantially reduce this cost since switching is performed directly on the digital bit stream and no costly analogue/digital conversion is needed. A combination of digital switching and transmission will therefore tend to lower the overall cost.

❑ *IC-technology.* Developments in integrated circuit technology seem to point to favourable cost levels and a high degree of reliabilty.

❑ *Integration of services.* As a digital medium a PCM-link can transmit not only speech but also data, telex, encoded visual information etc. A PCM-channel has a capacity of 64,000 bit/s which makes it a very powerful data channel.

New transmission media. Future wide-band transmission media, such as waveguides and optical fibres are more suitable for digital than analogue transmission.

The project consists of 3 chapters;

Chapter 1 presents five major principles: sampling, quantizing, electrical representation of PCM signals, Coding and Companding, demodulation for PCM systems. This theoretical material is used it be representation first order PCM system. The functional and timing diagram is system are presented.

Chapter 2 starts with description 30 and 24 channels standard PCM systems rdecommended by CCIT. The more important accept of PCM modulation technique-digital multiplexing hierarchy is analysed. The evaluation of Transmission rate for different level hierarchy are clearly calculated and explained.

Chapter 3 includes an application PCM system in the public switched telephone network. Different types of network is considered. New method of switching is given to interconnect ceutial office through the use of interoffice trunks and tudem trunus. Outside the local area told trunus and intertool trunus are considered.

The text is arrauged so that it can be used by students to study same part of the subject "Telecommunication". It can be useful for engineers too.

CHAPTER I W

THE FUNCTION BLOCK OF THE PCM SYSTEM

The statement above gives some idea about the basic processes in pulse code modulation. Here we shall give these processes their right names.

The process of choosing measuring points on the analoguee speech curve is called sampling. The measurement values are called samples. When samlpling, we take the first step towards a digital representation of the speech signal as the chosen sampling instants give us the time coordinates of the measuring points.

The amplitudes of the samples can assume each value in the amplitude range of the speech signal. When measuring the sample amplitudes we have to round off for practical reasons. In the rounding-off process, or the quantizing process, all sample amplitudes between two marks on the scale will be given the same quantized value. The number of quantized samples is discrete as we have only a discrete number of marks on our scale.

Each quantized sample is then represented by the number of the scale mark, i.e. we know now the coordinates on the amplitude axis of the samples.

The processes of sampling and quantizing yield a digital representation of the original speech signal, but not in a form best suited to transmission over a line or radio path. Translation to a different form of signal is required. This process is known as encoding. Most often the sample values are encoded to binary form, so that each sample value is represented by a group of binary elements. Typically, a quantized sample can assume one of 256 values. In binary form the sample will be represented by a group of 8 elements. This group is in the following called a PCM word. For transmission purposes the binary values 0 and I can be taken as corresponding to the absence and presence of an electrical pulse.

On the transmission line the pulses in the PCM words will become gradually more distorted. However, as long as it is possible to distinguish between the absence and the presence of a pulse, no information loss has occurred. If the pulse train is regenerated, i.e. badly distorted pulses are replaced by fresh pulses at suitable intervals, the information can be transmitted long distances with practically no distortion at all. This is one of the advantages of digital transmission over analogue transmission; the information is contained in the existence or not of a pulse rather than in the form of the pulse.

In our picture of the graph and the table this is analogous to the fact that the information in the table is not affected if the digits are badly written as long as they are legible. But if the graph is badly drawn, loss of information is inevitable.

On the receiving side the PCM words are decoded. i.e. they are translated back to quantized samples. The analogue speech signal is then reconstructed by interpolation between the quantized samples. There is a small difference between the analogue speech signal on the receiving side and the corresponding signal on the transmitting side due to the rounding off of the speech samples. This difference is known as quantizing distortion.

The function blocks in the pulse code modulation process are shown in figure 1.1.

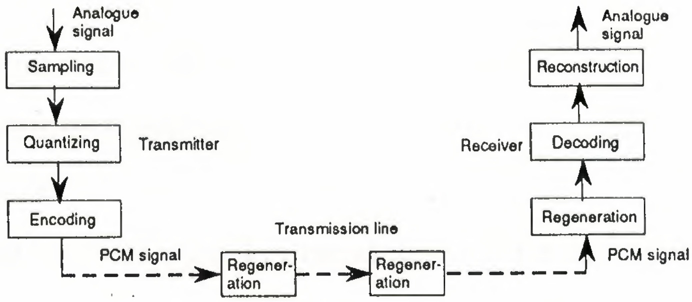

Figure 1 - 1. *Pulse code modulation Function blocks*

1.1. Sampling

In the practical electrical meaning, to sample is to take instaneous values of the analogue signal at equal time intervals. See figure 1-2.

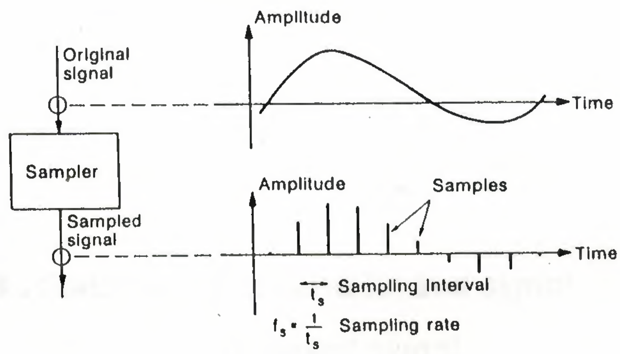

Figure 1 - 2. *The sampling process*

The sampled signal is a train of pulses, whose envelope is the original signal.

Now, what should be thesampling rate, i.e. the number of sampllles per second? The answer to this question is given by the Sampling Theorem, which also illustrates the fundamental fact that the information contained in the signal is not affected by sampling:

The sampled signal contains within it all information about the original signal if:

- ❑ the original signal is band limited, i.e. it has no frequency components in its spectrum beyond some frequency B
- ❑ the sampling rate is equal to or greater than twice B, i.e. $f_S \geq 2B$.

The sampling theorem is illustrated in figure I-3. Obviously, the spectrum of the sampled signal contains the spectrum of the original signal, i.e. no information loss has occurred.

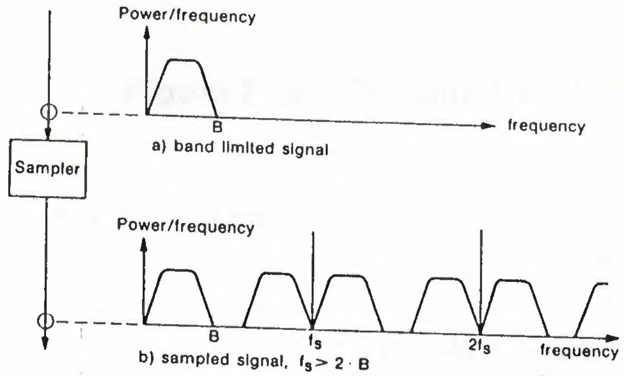

***Figure 1 - 3** . Spectrum of a) band-limited signal*

b) sampled signal

In telephony, the part of the speech spectrum between 300 and 3400 Hz is used. The human speech spectrum extends from a lowest frequency of some 100 Hz up to very high audio frequencies. The telephone set reduces this frequency range, but not enough at high frequencies so in order to come below this band limit at 3400 Hz, the speech signal must be low-pass-filtered before sampling.

plitude. This means that a loud talker and quiet talker let a listener hear the same quantizing distortion. Relative to the speech levels, the quiet talker generates much more distortion than the loud talker. Furthermore, a statistical analysis shows that for an individual talker, small amplitudes are much more probable than large ones.

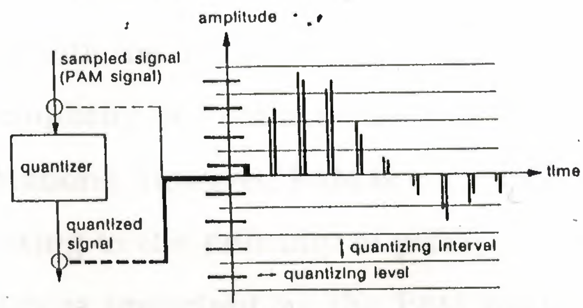

Figure I - 4 . *The quantizing process*

1.3. Coding & Companding

Practical PC systems use seven-and eight-level binary codes, or

$$2^7 = 128 \text{ quantum steps}$$

$$2^8 = 256 \text{ quantum steps}$$

Two methods are used to reduce the quantum steps to 128 or 256 without sacrificing fidelity. These are nonuniform quantizing steps and companding before quantizing, followed by uniform quantizing. unlike data transmission, in speech transmission there is a much greater likelihood of encountering signals of small amplitudes than those of larger amplitudes.

A secondary but equally important aspect is that coded signals are designed to convey maximum information, considering that all quantium steps (meanings or characters) will have an equally probable occurrence

A sampling rate of 8000 Hz is used for PCM systems in telephony. This rate is somewhat higher than twice the highest frequency in the band, 3400 Hz, due to difficulties in making low-pass filters steep enough.

The sampled signal is often said to be pulse amplitude modulated as it consists of a train of pulses, whose amplitudes have been modulated by the original signal. Pulse Amplitude Modulation (PAM) is an analogue pulse modulation method as the amplitudes of the pulses may vary continously in accordance with the original signal variations.

The relative simplicity of PAM systems makes them attractive for some telephony applications. However, PAM is unsuitable for transmission over long distances owing to the difficulty of pulse regeneration with sufficient accuracy, which is important as the PAM pulses contain the information, in the pulse form.

1.2. Quantizing

The continuous pulse amplitude range is broken down to a finite number of amplitude values in the quantizing process. The aplitude values in the quantizing process. The amplitude range is divided into intervals, and all samples whose amplitudes fall into one specific quantizing interval are given the same output amplitude. See figure I-4. The rounding off of the samples causes an irretrievable error, quanting distortion, in the signal.

This voluntary sacrifice, which can be brought down to suitable low limits by making the number of permitted amplitude levels large enough, is accepted because it makes error-free transmission possible by only having a discrete number of amplitudes.

In figure I-4 the quantizing distortion is independent of sample am-

(i.e., the signal-level amplitude is assumed to follow a uniform probability distribution between 0 and ± the maximum voltage of the channel). To circumvent the problem of nonequiprobability of signal level for voice signals, specifically, that lower - level signals are more probable than higher-level signals, larger quantum steps are used for the larger-amplitude portion of the signal, and finer steps are used for the signals with low amplitudes. The two methods of reducing the total number of quantum steps can now be more precisely labeled:

* Nonuiform quantizing performed in the coding process.
* Companding (compression) before the signals enter the coder, which now performs uniform quanttizing on the resulting signal before coding. At the receive end, expansion is carried out after decoding.

Most practical PCM systems use complanding to give finer granularity (more stepss) to the smaller amplitude signals. This is instantaneous companding, as compared to the syllabic companding used in analog carrier telephony. Compression imparts more gain to lower amplitude signals. The compression and later expansion functions are logarithmic and follow one of two laws, the A law or the "mu" (μ)law. The curve for the A law may be plotted from the formula

$$\left(\frac{A|x|}{1+\ln(A)}\right) x \le |x| \le \frac{1}{A}$$
$$\left(\frac{1+\ln|x|}{1+\ln(A)}\right) \frac{1}{A} \le |x| \le 1$$

where A = 87.6. The curve for the μ law may be plotted from the formula:

$$Y = \frac{\ln(1+\mu|x|}{\ln(1+\mu)}$$

where x is signal imput amplitude and μ = 100 for the original North American T1 system (now out dated) and 255 for later North American (DSI) systems and the CCITT 24-channel system.

A common expression used in dealing with the "quality" of aPCM signal is the signal-to-distortion ratio (expressed in decibels). Parameters A and μ determine the range over which the signal-to-distortion ratio is comparatively constant. This is the dynamic range. Using a μ of 100 can provide a dynamic range of 40 dB of relative linearity in the signal-to-distortion ratio.

In actual PCM systems the companding circuitry does not provide an exact replica of the logarithmic curves shown. The circuitry produces approximate equivalents using a segmented curve, and each segment is linear. The more segments the curve has, the more it approaches the true logarithmic curve desired. Such a segmented curve is shown in Figure I-5. If μ law were implemented using a seven (height)-segment linear approximate eqivalent, it would appear as shown in Figure I-5. Thus on coding, the first three coded digits would indicate the segment number (e.g. $2^3 = 8$). Of the seven-digit code, the remaining four digits would divide each segment into 16 equal parts to identify further the exact quantum step (e.g., $2^4 = 16$). For small signals, the companding improvement

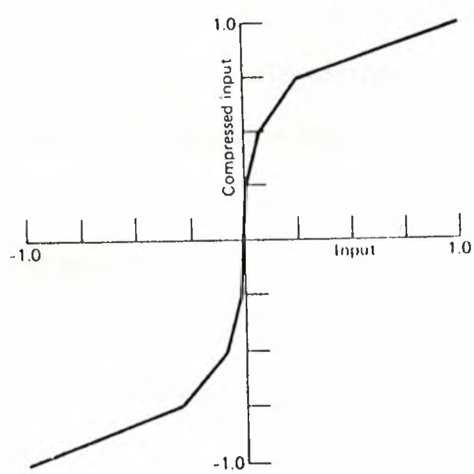

Figure I - 5. *Seven-segment linear approximate of the logarithmic curve for μ law (μ = 100)* is approximately

A law : 24 dB

μ law : 30 dB

using a seven-level code. These values derive from the equation of companding improvement or

$$G_{dB} = \frac{\text{Uniform (linear) scale}}{\text{Companded scale}}$$

Coding in PCM systems utilizes straighforward binary codes. Examples of such coding are shown in Figure I-5a, which is expanded in Figure 9.7, and in Figure 9.8, which is expanded in Figure 1.6.b showing a number of example code levels.

The coding process is closely related to quantizing. In practical systems, whether the A law or the μ law is used, quantizing employes segmented equivalents of the companding curve (Figures I-6 and1-8), as discussed earlier. Such segmenting is a handy aid to coding. Consider the European 30 + 2 PCM system, which uses a 13-segment approximation of the A-law curve (Figure 1-6). The first code element indicates whether the quantum step is in the negative or positive half of the curve. For example, if the first code element were a 1, it would indicate a positive value (e.g., the quantum step is located above the origin). The following three-code elements (bits) identify the segment, as there are seven segments above and seven segments below the origin (horizontal axis).

The first four elements of the fourth + segment are 1101. The first 1 indicates it is above the horizontal axis (e.g., it is positive). The next three elements indicate the fourth step or

0-1000 and 1001

1-1010

2-1011

→ 3-1100

4 -1101

5-1110 etc.

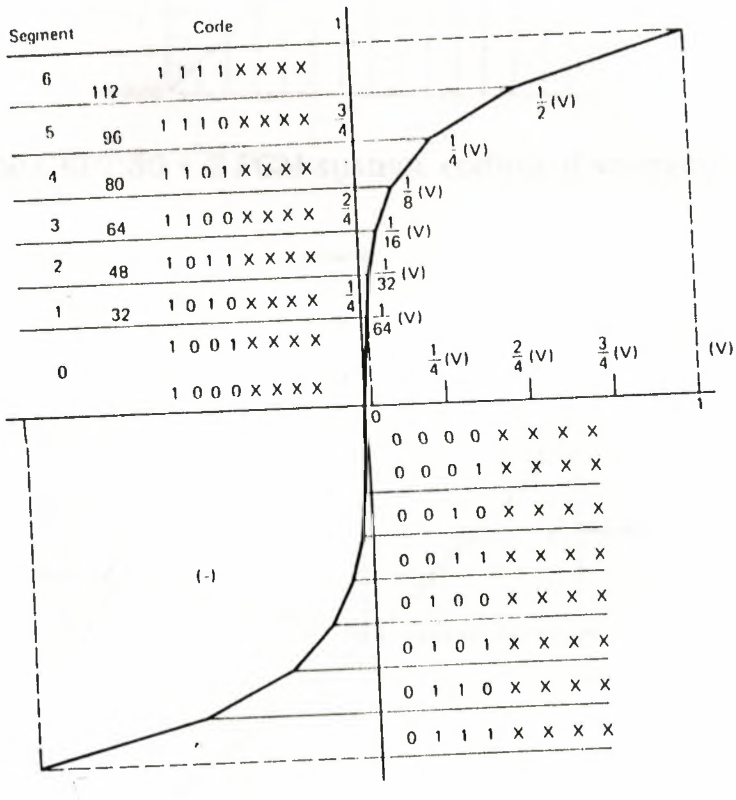

Figure I - 6. *Quantization and coding used in the CEPT 30 + 2 PCM system.*

Figure 1.7 shows a "blowup" of the uniform quantizing and subsequent straightforward binary coding of step 4. This is the final segment coding, the last four bits of a PCM code word for this system. Note the 16 steps in the segment, which are uniform in size.

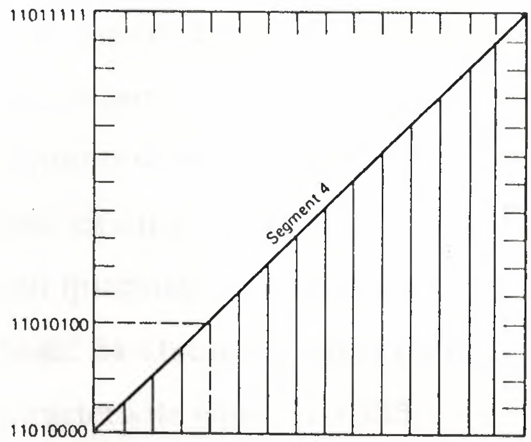

Figure I - 7. *The CEPT 30 + 2 PCM system, coding of segment (4 positive).*

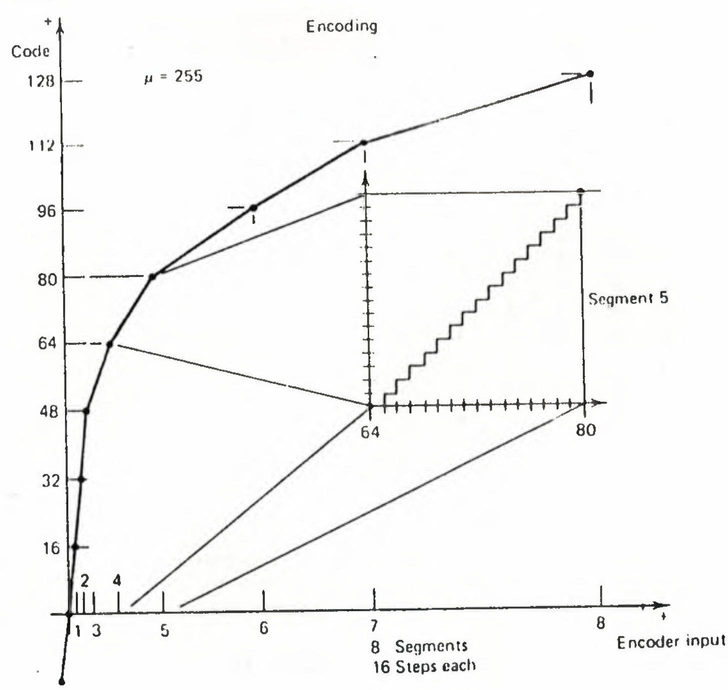

Figure I - 8. *Positive portion of segmented approximation of μ law quantizing curve used in North American (ATT) DS1 PCM channelizing equipment. Courtesy of ITT Telecommunications, Raleigh, N.C.*

The North American DSI PCM system uses a 15-segment approximation of the logarithmic μ law. Again, there ase actually 16 segments. The segments cutting the origin are colinear and counted as one. The quantization in the DSI system is shown in Figure 1-8 for the positive portion of the curve. Segment 5, representing quantizing steps 64 through 80, is shown blown up in Figure 1-8, Figure 1-9 shows the DSI coding. As can be seen in the figure, again the first code element, whether a 1 or a 0, indicates whether the quantum step is above or below the horizontal axis. The next three elements identify the segment, and the last four elements (bits) identify the actual quantum level inside the segment. Of course, we see that the DSI is a basic 24-channel system using eight-level coding with μ-law quantization characteristic where μ = 255.

Code Level		Digit Number							
		1	2	3	4	5	6	7	8
255	(Peak positive level)	1	0	0	0	0	0	0	0
239		1	0	0	1	0	0	0	0
223		1	0	1	0	0	0	0	0
207		1	0	1	1	0	0	0	0
191		1	1	0	0	0	0	0	0
175		1	1	0	1	0	0	0	0
159		1	1	1	0	0	0	0	0
143		1	1	1	1	0	0	0	0
127	(Center levels)	1	1	1	1	1	1	1	1
126	(Nominal zero)	0	1	1	1	1	1	1	1
111		0	1	1	1	0	0	0	0
95		0	1	1	0	0	0	0	0
79		0	1	0	1	0	0	0	0
63		0	1	0	0	0	0	0	0
47		0	0	1	1	0	0	0	0
31		0	0	1	0	0	0	0	0
15		0	0	0	1	0	0	0	0
2		0	0	0	0	0	0	1	1
1		0	0	0	0	0	0	1	0
0	(Peak negative level)	0	0	0	0	0	0	1*	0

*One digit is added to ensure that the timing content of the transmitted pattern is maintained.

Figure I - 9. *Eight-level coding of North American (ATT) DS1 PCM system. Note that there are actually only 255 quantizing steps because steps 0 and 1 use the same bit sequence, thus avoiding a code sequence with no transitions (i.e., 0's only).*

As we know, pulses with two levels, i.e. binary pulses, are attractive for transmission as they are easy to regenerate on the transmission line. It is not dfficult to build regenerator circuits able to determine whether a pulse is present or not.

Present-day practical system use binary encoding of the quantized speech samples. See figure 10. As telephony uses 256 quantizing levels, each sample will be encoded to a code group, or PCM word, consisting of 8 binary pulses (8 bits).

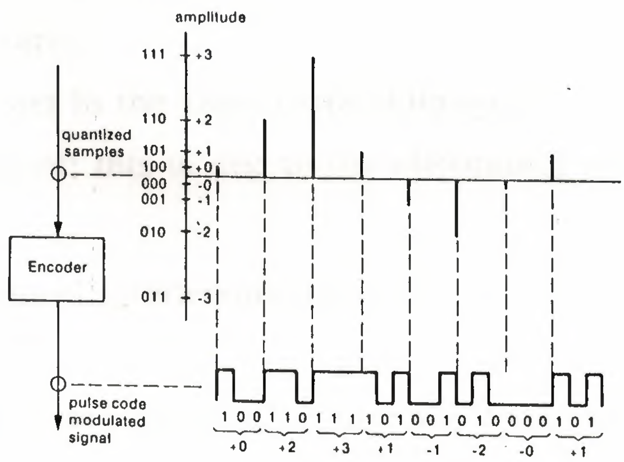

Figure 1- 10. *Encoding of quantized samples with 8 quantizing levels (3 binary digits/code word).*

As the sampling rate used is 8000 samples/second, one pulse code modulated speech signal will generate a 64 kbit/s digital signal.

1.4. Electrical Representation of PCM signals

Digital signals within the terminal are usually transmitted in the form of a unipolar pulse train in the nonreturn-to-zero (NRZ) mode, see figure 11. This signal form is not appropriate for transmission over long distances.

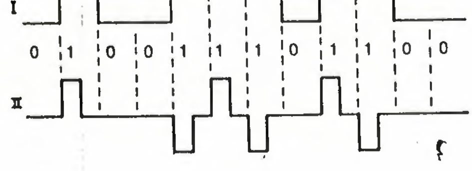

Figure 1- 11. *Binary information represented in:*

I a unipolar nonreturn-to-zero (NRZ) pulse train.

II a bipolar return-to-zero (RZ) pulse train.

A better form is a bipolar return-to-zero (RZ) signal. The advanteges of this signal are :

☆ it has no power in the lower parts of its spectrum, i.e. it has no direct current component; this is due to the alternating polarities of the pulses

☆ the intersymbol interference is reduced by the return-to-zero feature.

Of course, even this signal will be attenuated and distorted during transmission, and noise will be added to it.

At some point on the transmission line the signal must be restored. This is done by inserting a device on the line that first examines the distorted pulse train to see whether the likely binary value is 1 or 0, and then generates and transmits to the line new pulses according to the result of the examination. Such a device is called a regenerative repeater. See figure 12.

At the same time as the pulses are reshaped, the noise added during transmission is eliminated at the least if the noise signal amplitude is not large enough to bring the received code signal to the wrong side of a regenerator decision level. Normally, the regenerated code signal is identical to the transmitted original code signal. Even after a large number of regenerative repeaters, the code signal is practically identical to the original signal. This is the reason for the high transmission quality that is obtainable with PCM transmission system.

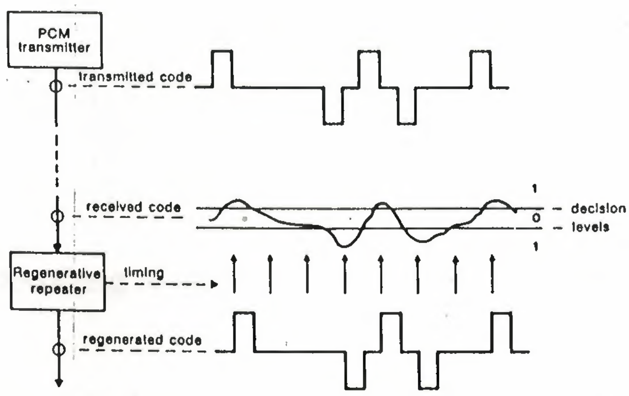

Figure 12. *Pulse forms on a transmission line.*

1.5. Demodulation

The processes in the receiver that convert the incoming PCM signal to an analogue speech signal again are regeneration, decoding and reconstruction.

The regeneration process has the same aim and is performed in the same way as on the transmission line, i.e. the distorted pulses are replaced by new square pulses, see figure 12. Before entering the decoder the bipolar signal is reconverted to unipolar. In the decoding process the code words generate amplitude pulses, whose heightts are the same as the heights of the quantized samples, which generated the code words. So after passing through the decoder the train of quantized samples is retrieved. See figure 13.

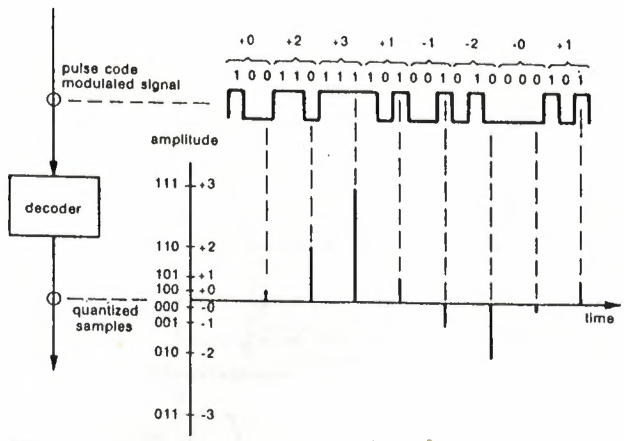

Figure 13. Decoding of encoded amplitude levels

The analogue signal is reconstructed in a low pass filter, figure 14a. This can be seen from figure 14b. The spectrum of a sampled signal contains the spectrum of the original signal as has been shown in figure I-3. A low pass filter with a cut-off frequency at B Hz takes away all frequency components in the spectrum above B Hz and the spectrum of the desired analogue signal is left.

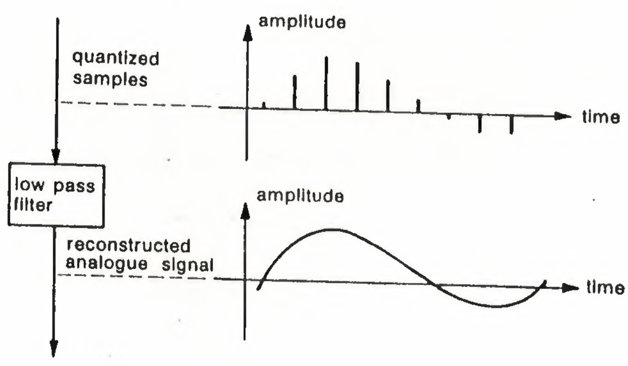

***Figure I - 14.a** Reconstruction of the analogue*

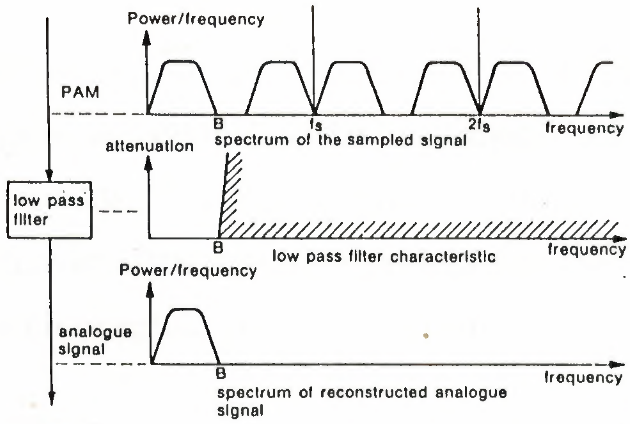

Figure I - 14b. *Reconstruction of the analogue signal shown by spectrum diagram.*

CHAPTER II

PCM TRANSMISSION SYSTEM

We have now dealt with the fundamental principles of PCM. In this part of the chapter we shall describe how these principles are used to build up practical PCM transmission systems. However, we shall start by explaining the time-division multiplex principle as this makes PCM transmission systems for telephony economically attractive.

2.1. First Order PCM Systems

Several signals in pulse from can use a common transmission path if the signals have different phases. Figure 2-1 shows how three PAM-signals are time-division multiplexed on the same transmission line. The pulses of the three signals are interleaved by opening the sampling gates one by one cyclically. During one cycle the transmission line receives one PAM pulse from each of the participating signals. Such a set of pulses is called one frame. The time interval that each of these pulses occupies is called a time slot. In this example each frame has three time slots.

On the receiving side the pulses are distributed again. This is done by opening the sampling gates cyclically in the same manner as on the transmitting side. Due regard must of course be paid to the transmission delay. This delay has been omitted in figure 2-1 for clarity.

In the case of PCM signals the time-division multiplexing is most ofter carried out before the samples are pulse coded, i.e., the samples from the participating analogue signals are combined on a common PAM transmission line. See figure 2-2.

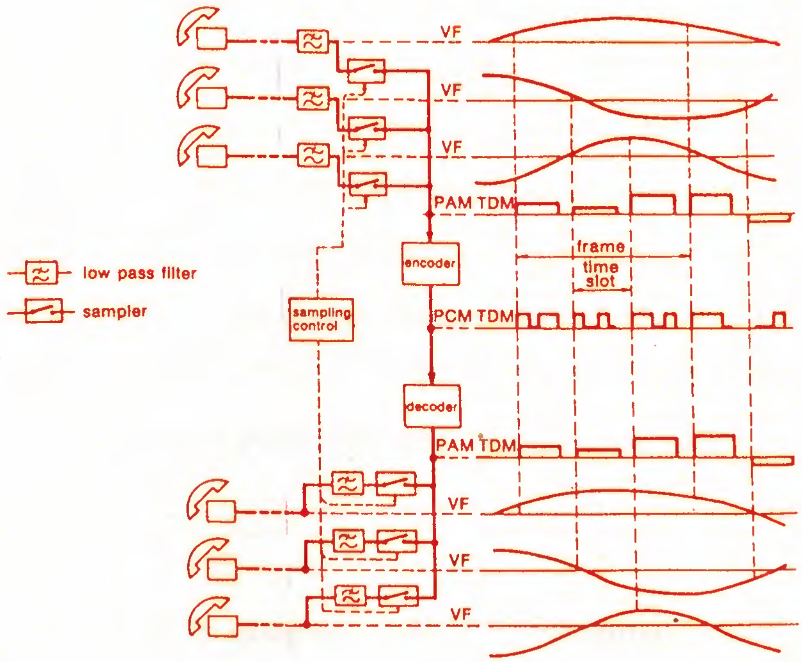

Figure 2-1. *A PAM transmission system using time-division multiplex (TDM)*

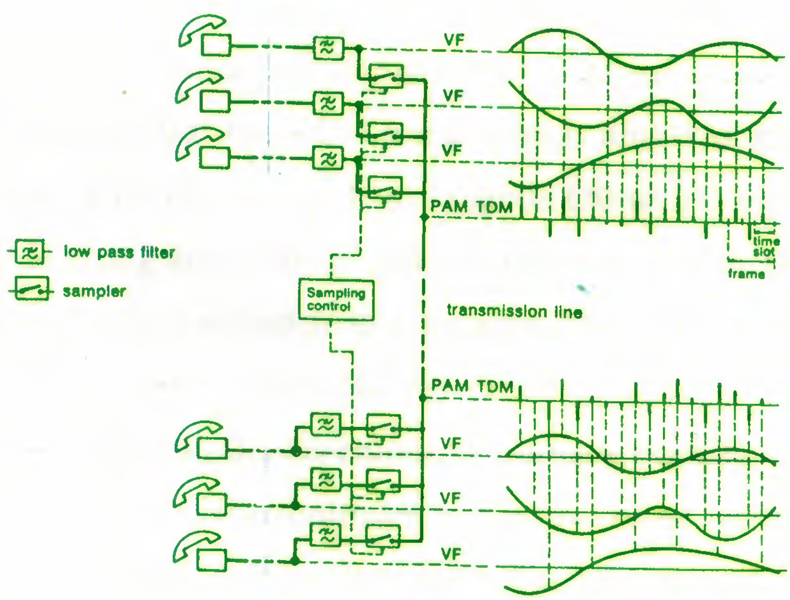

Figure 2-2. *A PCM-TDM transmission system. Attenuation and delay on the PAM and PCM transmission lines are not shown.*

In this way the coding equipment can be used in time-division multiplex. We see from the figure that the PCM pulses are not nterleaved pulse by pulse, but PCM word by PCM word. This is often called time slot interleaving. PCM systems used in telephony are most often TDM systems, so when we read or hear the term "PCM system" it is almost always referring to a PCM-TDM system. However it must not be forgotten that PCM in itself can be used, and is used in some cases, on a one-channel basis.

2.2. First Order PCM System Recommended by CCITT Frame Structures

As mentioned in chapter I, CCITT recommends two first order, or primary, PCM systems for use in telephony; the 30-channel system, proposed by CEPT, and the 24-channel system proposed by AT & T. The first order systems will form the basis for hierarchies of digital transmission systems.

We have to distinguish between the PCM multiplex equipment, or the PCM terminal, and the/PCM transmission line. The multiplex equipment converts a number of analogue signals (30 or 24) to a digital signal on the transmitting side and carries out the inverse functions on the receiving side. The transmission line conveys the digital signals between two multiplex equipment units. See figure 2-3.

In the following, the 30-channel multiplex equipment will be treated in some detail, as this multiplex forms the basis for the subsequent presentation of digital telephony in this book. A summary of the most important data on the 24-channel multiplex is also given. The presentation of first order PCM systems will end with a brief description of the transmission lines.

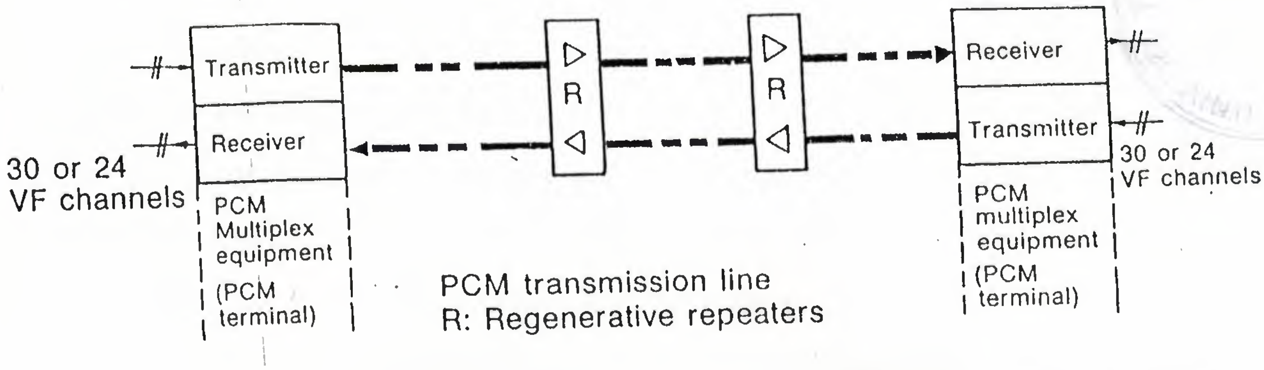

Figure 2-3. *Two multiplex equipment units.*

Thirty analogue speech channels together with associated signalling are converted to one digital signal by the 30-channel system. The structure of this digital signal is shown in figure 2-4.

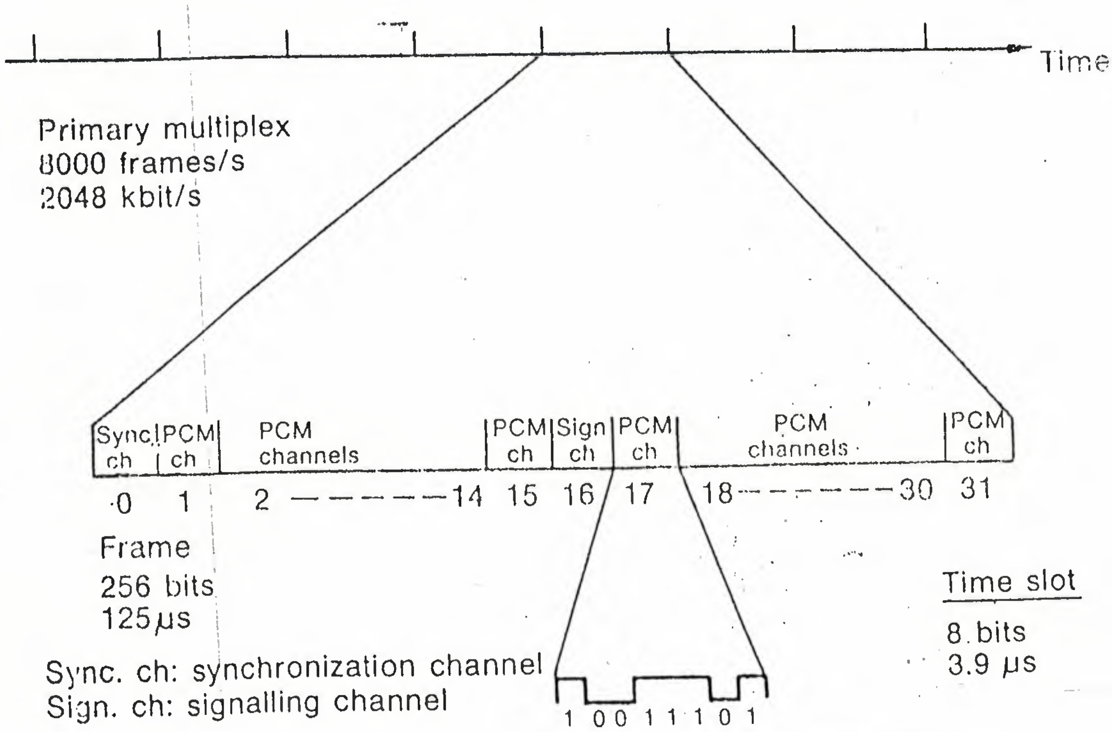

Figure 2-4. *Frame structure of the 30-channel primary multiplex.*

The digital signal is divided into frames, with a repetition rate of 8000 frames/sec. This is of course because the sampling frequency is 8000 Hz and the fact that the frame contains one binary coded sample from each of the analogue signals. Each frame consists of 32 eight bit time slots. Of these, 30 time slots are used for PCM channels and the remaining two for synchronization and signalling.

The PCM channels carry analogue signals within the frequency band 300-3400 Hz, coded according to the A-law, see figure 1-5.

The synchronization time slot, time slot 0 in each frame, contains 8 bits whose purpose is to form a recognition signal to the receiver in order to keep this synchronized to the transmitter so that each PCM channel can be correctly identified. This function is the same as the function indicated by the sampling control block in figures 2-1 and 2-2.

The signalling time slot, No.16, can be used in many ways. The great signalling capacity, 64 kbit/s, offers flexibility in chosing suitable schemes for different purposes. This is important when considering the digital network of the future.

CCITT has recommended the use of the signalling time slot for either common channel or channel associated signalling. The arrangements for common channel signalling are not yet specified, so here we can only go into details concerning the channel associated signalling scheme.

The signalling scheme is the one used today when introducing PCM primary systems into the existing network. The scheme uses the time slots 16 in sequences of 16 frames, referred to as multiframes, as shown in to figure 2-5.

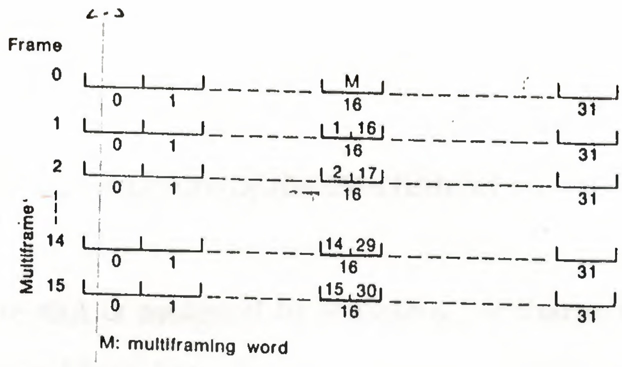

Figure 2-5. *Structure of the channel associated signalling scheme for the 30-channel PCM system.*

In the first frame of tthe sequence, frame 0, time slot 16 carries a multiframing word, i.e. a recognition signal that tells the receiver that a new multiframe has started. The eight bits of time slot 16 in the next frame, frame 1, are divided so that the first four bits carry signalling information associated with PCM channel 1, and the last four bits carry signalling information associated with PCM channel 16. In frame 2, time slot 16 carries signalling information associated with PCM channel 16. In frame 2, time slot 16 carries signalling information for the channels 2 and 17 and so on up to frame 15, the last frame in the multiframe, which carries signalling information for the channels 15 and 30. Then the next frame is frame 0 in the next multiframe.

TThus four signalling bits are associated with each PCM channel. Each bit can be used to reproduce the state of a signalling relay in a junctor connected to the PCM terminal, i.e. the scheme provides four signalling channels per PCM channel; Normally, for conveying conventional signals, only one or two of the channels are used.

The 24-channel PCM multiplex has a somewhat different structure as can be seen from figure 2-6.

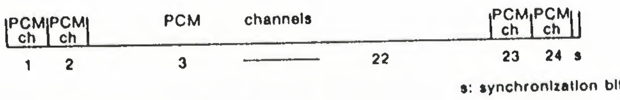

Figure 2-6 *Frame structure of the 24-channel system.*

No special time slot is assigned to signalling. A channel associated signalling scheme is achieved by taking the least significant bit in every PCM channel for signalling purposes every 6th frame. For synchronization one extra bit is inserted in the frame. This bit can also be used for common channel signalling.

A summary of important technical data for the primary multiplexes is shown in figure 2-7. It is obvious that the primary multiplexes are not compatible; they have, for example, a different number of time slots and different signalling possibilities. Not even the time slots are compatible as the systems use different encoding laws. However, work is going on at CCITT aimed at finding a digital method for conversion between PCM words using different encoding laws in order to avoid conversions to analogue when connecting different PCM systems to each other.

	the 30-channel system	the 24-channel system
audio frequency band	300 -3400 Hz	300 -3400 Hz
sampling rate	8000 Hz	8000 Hz
bits/sample	8	8*
time slots/frame	32	24
PCM channels/frame	30	24
output bit rate	2048 kbit/s	1544 kbit/s
encoding law	A-law, A=87.6	μ -law, μ =255
signalling capacity		
channel associated signalling	1-4 sign. ch./PCM ch.	1-2 sign. ch./PCM ch.
common channel signalling	64 kbit/s	4 kbit/s

*) only 7 bits each 6th frame if channel associated signalling is used.

Figure 2-7 *Technical data for the primary multiplexes.*

The PCM transmission lines used for interconnecting primary multiplexes are most often already existing pair cables used for analogue voice frequency transmission. For a PCM line we need two pairs, one for each direction see figure 2-3. The line must be equipped with regenerative repeaters every 1.5-2.5 km, depending on the type of cable. This distance is about the same as the distance between loading coils on loaded cable circuits, and conversion of loaded pairs into PCM transmission lines can be carried out merely by replacing the loading coils with PCM regenerative repeaters on specially selected pairs.

The transmission lines are designed to convey digital signals of specified rates 2048 or 1544 kbit/s. The digitalsignals must fulfil requirements concerning not only pulse polarities, see figure 1-7, but also pulse distribution. The timing circuits in the repeaters need, for their operation, digital signals with alternating pulses. Long sequences of zero-bits must be avoided. This is done by a special line code that converts some of the zero-bits to pulses in such a way that these extra bits can be removed before arriving at the decoder.

2.3. Second Order PCM System

The primary PCM systems are intended for short-distance applications. In the medium and long distance network, where high channel capacity is demanded, it is more economical and practical to group together larger number of PCM channels to one common transmission line, thus forming higher order systems, than to use several primary PCM systems.

In general, multiplexes can be of two types:

✔ PCM multiplexes drive one sngle digital signal from a number of analogue signals by a combination of pulse code modulation and time division multiplexing and also carry out the inverse function. For example, the first order multiplexes decribed earlier are of this type.

✔ Digital multiplexes derive one single digital signal by combining a number of digital signals by time division multiplexing and also carry out the inverse function.

Digital transmission lines convey dgital signals between multiplex equipment units. These lines are designed to carry digital signals of specified rates, but they are not dependent on what type of original signal is conveyed in digital form. TThat is, the same transmission line can be used for both PCM multiplexes and digital multiplexes if these have the same rates on the multiplexed digital signals. Even other types of digital signals, for example digitally coded visual telephone signals or data signals, can use the transmission lines if their digit rates are correct.

2.4. TDM Hierarchy

Two second order systems have been recommended by CCITT. These have digital multiplexes and are based on each of the two primary multiplexes. They both combine four primary PCM signals to one digital signal. See figure 2-8.

The signals are multiplexed by bit interleaving, that is the participating signals are combined bit by bit, see figure 2-9. This is more practical than time slot interleaving, in which the participatting signals are combined time slot by time slot, as in this latter case it is necessary to collect the bits of the time slots in buffers before interleaving can be carried out.

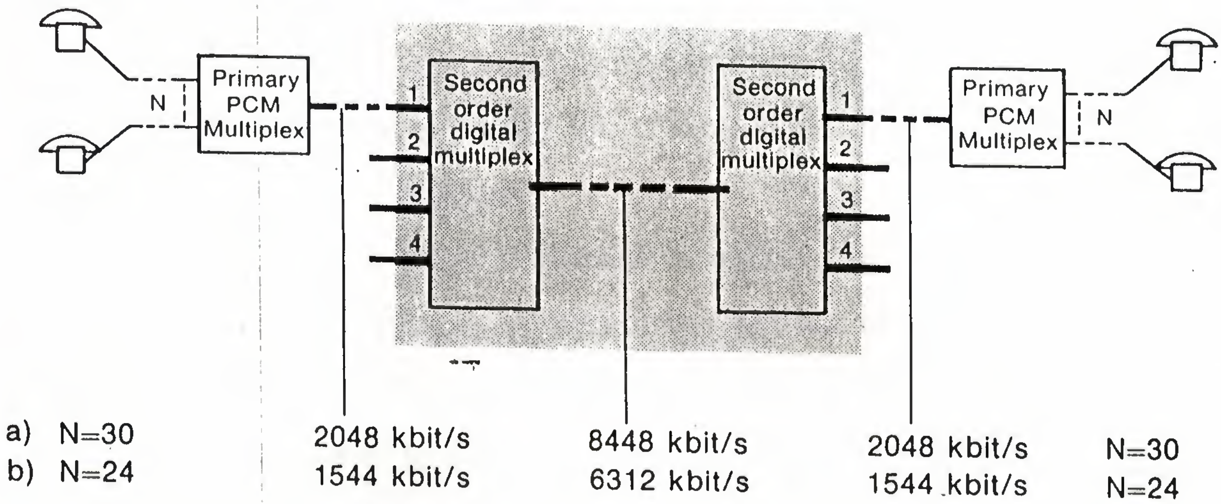

Figure 2-8. *Second order digital multiplexing of primary PCM signals. (a) The CEPT scheme. (b) The AT & T scheme.*

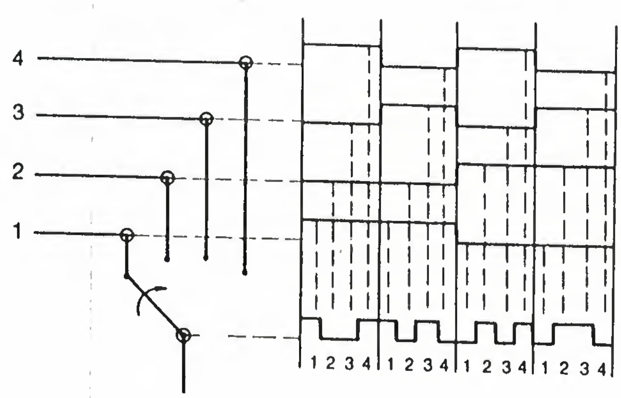

Figure 2-9. *Bit interleaving*

The digital multiplexes must accept that the primary signals, for practical reasons, have bit rates slightly differing from the ideal bit rate. In the systems recommended by the CCITT this is accomplished by having the second order bit rates somewhat higher than four times the ideal primary bit rates, thereby ensuring thatt, even "fast" primary multiplexes can be treated in a proper ways.

A second order PCM multiplex is already in use in the United States. This system combines 96 analogue channels to a 6312 kbit/s digital signal, i.e. it uses the same transmission line as the digittal multiplex according to figure 2-8. In Europe a second order PCM multiplex based on the parameters of the 30-channel system and the 8448 kbit/s transmission line has been discussed. This system is proposed to be time slot interleaved with 132 time slots. Of these, 128 can be used for speech channels, 2 for synchronization and 2 for signalling.

2.4.1. TDM Hierarchy Based on 30 Channel System

The digital transmission systems can be tied together in a hierarchy in the same way as the FDM systems. Two hierarchies are discussed, one based on the 24-channel system and the other based on the 30-channel system. One possibility for the latter is shown in figure 2-10.

From the figure, it can be seen that the transmission facilities are planned to be used not only by pulse code modulated speech but possible digit rates and ttransmission media are shown in figure 2.11.

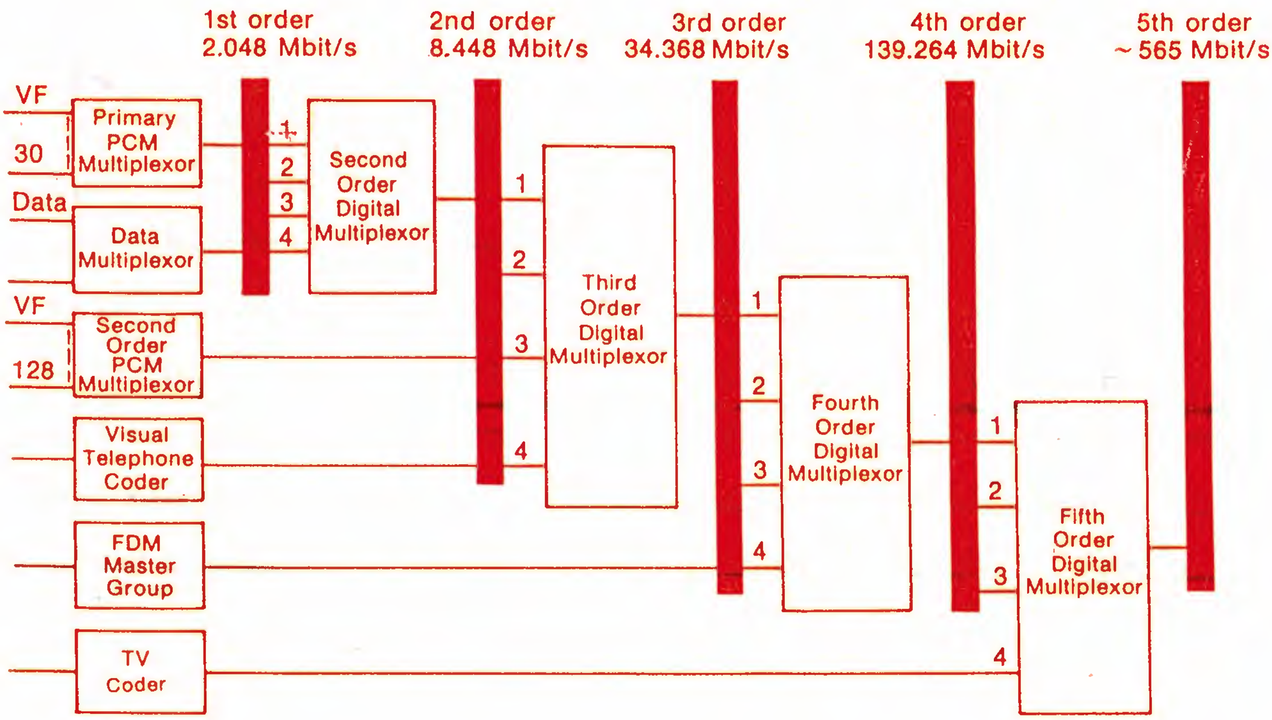

Figure 2-10. *A possible digital transmission hierarchy based on the 30-channel PCM system.*

Order	Digit rate Mbit/s	No. of speech channels for digital multiplexes	Pair cable	Pair cable new type	Micro coaxial cable	Small coaxial cable	Normal coaxial cable	Radio link, 12 GHz	Waveguide 30-120 GHz	Optical waveguides
1	2.048	30	×	×						
2	8.448	120		×	×	×		×		×
3	34.368	480		×	×	×	×	×		×
4	139.264	1920				×	×	×	×	×
5	~565	7680					×		×	×

Figure 2-11. *Digit rates and transmission media for a hierarchy based on the 30-channel PCM system.*

2.4.2. TDM Hierarchy Based on 24-Channel System

To take further advantage of the merits of TDM and digital transmission, the common carriers employ a hierarchy of further multiplexing as shown in Figure 2-12. FourT1 lines are multiplexed in an M12 multiplexer to generate a T2 transmission system, seven T2 lines convert to a T3 line in an M23 multiplexer and six T3 lines convert to a T4 line in an M34 multiplexer. At each stage additional frame synchronizing bits must be added as with the first order multiplexing so that at each multiplexer output it would be possible to distinguish which bits belong to which input.

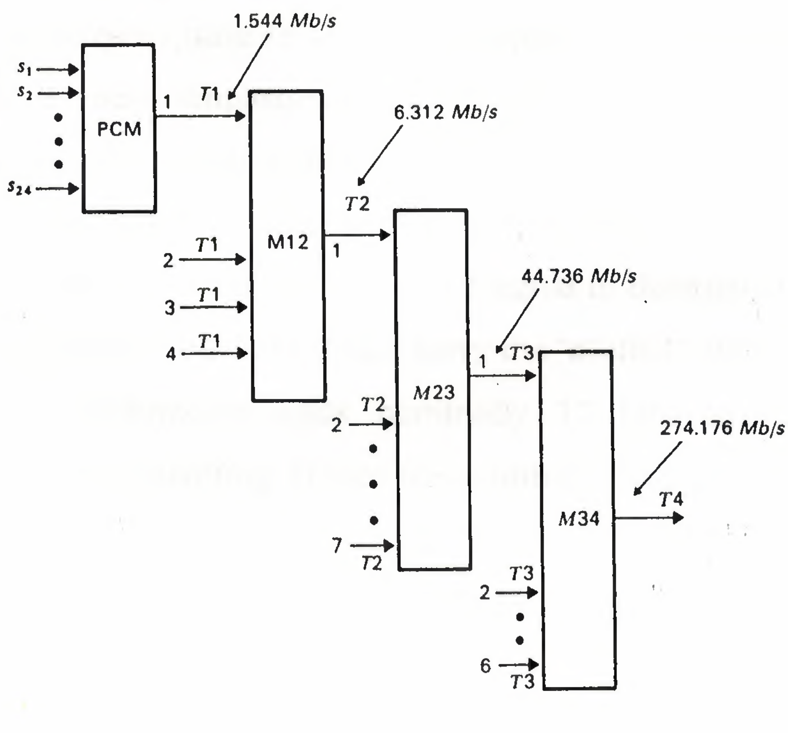

Figure 2-12. *TDM hiararchy.*

There is a problem that arises in connection with the higher orders of multiplexing that does not occur at the first order. In the first order multiplexing there is just a single clock to contend with, that is the clock that drives the commutator. However, the four input lines in the M12 multiplexer come from physically widely separated location and employ four separate unsynchronized clocks. These clocks are very stable crystal controlled oscillators which are, of course, set to operate at the same nominal frequency as nearly as possible. Since, however, chronized and will hence experience a relative frequency drift. Design specifications allow a drift from the nominal set frequency of ± 130 parts/million. As may be verified, if then two clocks have frequencies which differ by 2 x 130 = 260 parts/million, the faster clock will have generated one more time slot than the slower clock in the course of just 20 frames. For proper interleaving of bits at the M12 level it is necessary that all input bit streams have or be made to appear to have the same rate. To put the matter most simplistically, the process of adjusting bit rates to make them equal involves adding bits to the slower bit stream in an operation referred to as "pulse stuffing." Further bits must then be added to all bit streams to allow the receiver of the composite signal to distinguish time slots which carry information from slots which carry the "stuffed" bits.

The M12 multiplexer adds nominally 17 bits for frame synchronization and pulse stuffing. Hence the number of bits per frame is

$$193 \times 4 + 17 = 789 \text{ bits/frame}$$

The T2 line bit rate is therefore

$$f_b(TT2) = 789 \text{ bits/frame} \times 8000 \text{ frames/s}$$
$$= 6.312 \text{ Mb/s}$$

The M23 multiplexer adds nominally 69 bits for synchronization and pulse stuffing; hence the number of bits per frame for a T3 line is

$$789 \times 7 + 69 = 5592 \text{ bits/frame}$$

and

$$f_b(T3) = 5592 \times 8000 = 44.736 \text{ Mb/s}$$

The M34 multiplexer adds nominally 720 bits for synchronization and pulse stuffing and therefore tthe T4 system has a bit rate

$$f_b(T4) = 274.176 \text{ Mb/s}$$

A detailed analysis of the architecture and operation of these digital systems it to be found in Ref.3.

CHAPTER III

THE TELEPHONE NETWORK

The invention of the telephone in 1876 led to an explosive outgrowth of engineering developments. These developments continue to thrive to this date. This proliferation, through what has become known today as "the largest industry in the world." has led to the existing informational era in which we live. Despite the Bell System divestiture (breakup) on January 1, 1983. The telecommunicattions industry sttill exists, employing millions of people. Divestiture has clearly established a distinction between the principal entities or segments of the telecommunications industry. Over 1400 independent telephone companies exist today, making up what has become known as the public switched telephone network (PSTN).

The PSTN has been dubbed the "world's most complex machine." It is truly one of the modern wonders of the world. In the United States alone, the PSTN interconnects hundreds of millions of telephones through the largest network of computers in the world. Despite its electrical and mechanical sophistication, even the most unskilled person can utilize its services. At the touch of a dial, any two people, virttually anywhere in the world, can communicate with each other in a matter of seconds. In this chapter we examine the basic structure of the PSTN and how it has evolved over the years. The most recent technological advances are introduced.

3.1. The Public Switched Telephone Network (PSTN)

Since the invention of the telephone, the PSTN has grown proportionately with the increased demands to communicate. Switching services beyond metropolitian areas were soon developed, increasing the size and complexity of the central office. New methods of switching were required to interconnect central offices through the use of interoffice trunks and tandem trunks. Figure 3-1 depicts how today's central offices are connected through the use of trunks. In largely populated areas, a tandem office is used to minimize the number of trunks that a call must be routed through to reach its destination. Outside the local area, toll trunks are used to connect the central office to toll centers. Toll centers may be located in adjacent citties outside the local area. To achieve even longer distances, toll centers are interconnected by inttertoll trunks as shown.

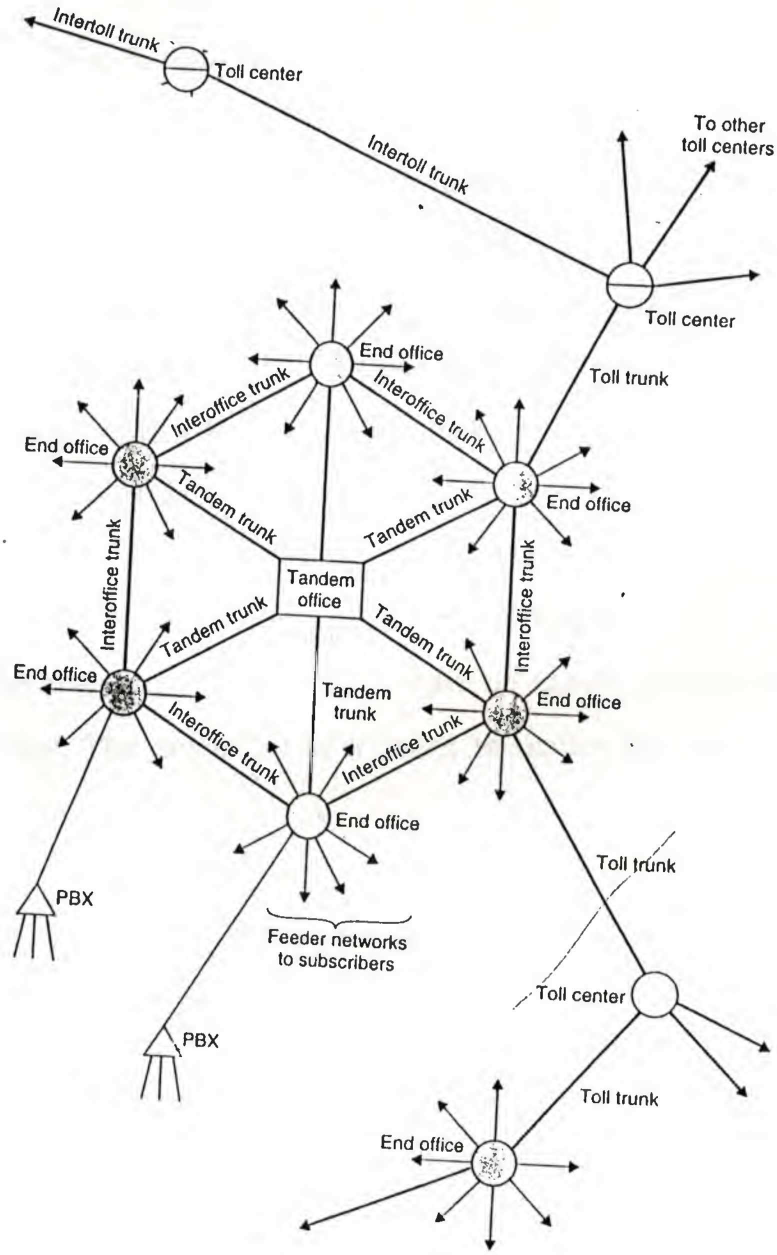

Figure 3-1 *Interconnection of switching exchanges in North America.*

3.1.1. Switching Hierarchy of North America

A hierarchy of switching exchanges evolved in North America to accommodate the demand for longer-distance connections. The PSTN has classified these exchanges into five levels of switching, as depicted in Table 3-1. At the lowest level, class 5, is the central office or end office. A large metropolis may require several end offices for service, whereas in a rural area a single end office is usually sufficient. When calls are made outside the local area, they are routed through class 4 centers, toll centers, and possibly higher levels of switching, depending on the destination of the call and the current traffic volume within the PSTN. To aid in the volume of traffic between toll centers, primary centers, sectional centers, and regional centers, classes 3,2, and 1, respectively, are used. Figure 3-2 illustrates the possible routes that a toll can take throughout the switching hierarchy. The best route is the shortest route or the route utilizing the smallest number of switching centers. A call may not always take this route, however. It depends on the availability of trunk circuits when the call is placed. Several alternative routes can be taken up and down the switching hierarchy. The selection of a route is under program control at the switching center.

TABLE 3-1. Hierarchy of Switching Exchanges in North America.

Class 1	Regional center	□
Class 2	Sectional center	△
Class 3	Primary center	○
Class 4	Toll center	⊖
Class 5	End office	◌

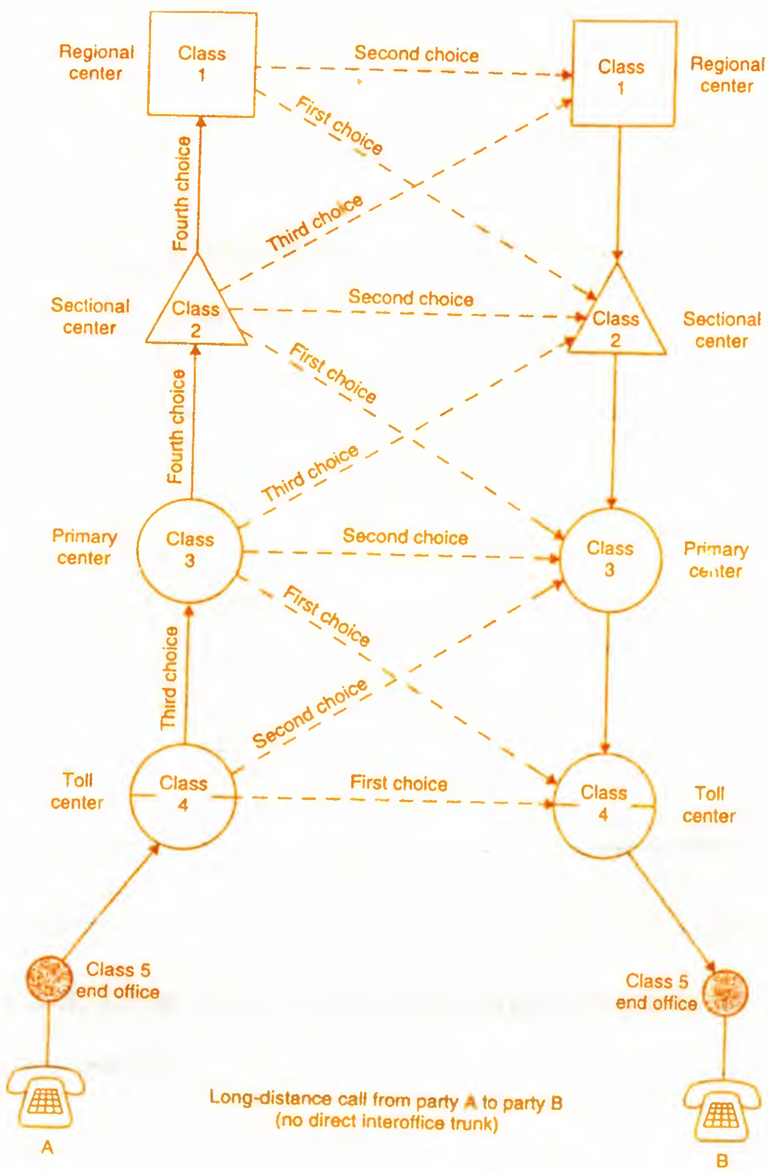

Figure 3-2. *Long-distance switching hierarchy within the PSTN depicting plossible routes to complete a call between parties A and B.*

Typically, trunks used to interconnect higher levels of switching are designed for high-speed transmission using the multiplexing techniques that will be described. These higher levels of switching make up the long-haul network. Calls placed onto the long-haul network are subject to toll fees. Figure 3-3 illustrates the long-haul network. To sustain higher transmission rates within the long-haul network, wideband trunk media are used. This includes coax, microwave ground and satellite links, and fiber-optic cables.

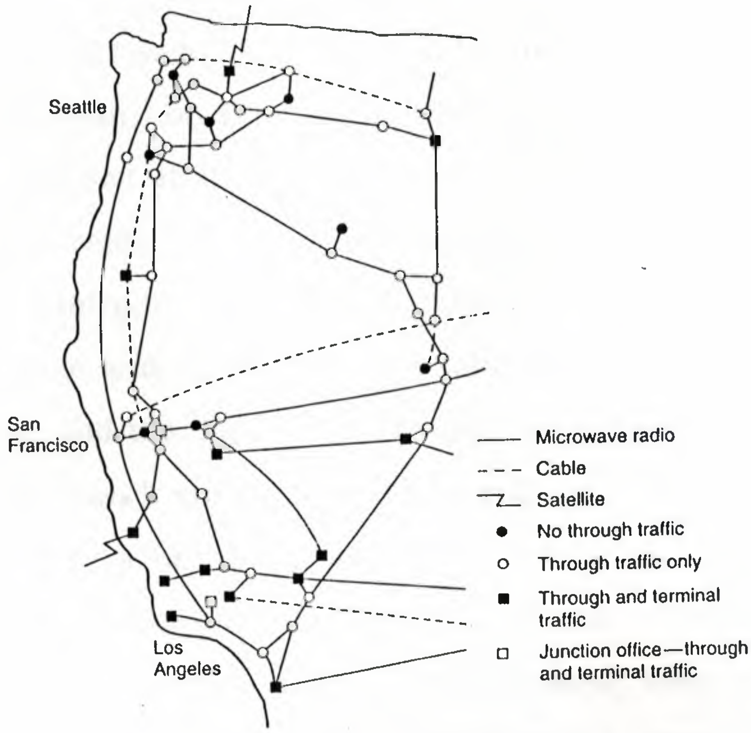

Figure 3-3. *Long-haul network depicting various trunk media. (Courtesy of Bell Laboratories.)*

3.2. Technical Consideratons in the Planning of Digital Networks for Multiexchange Areas

The use of remote concentrators to connect subscribers will make it economical to increase the exchange areas. Further the digital group selectors will have a capacity to handle the traffic from a large number of subcribers. The combination of digital group selectors and common channel signalling will make the use of both way circuits between the exchanges more attractive. Consequently the digital network will contain larger exchanges with more direct routes and there will be fewer tandems in the local network. But the same routing alternatives remain; direct routes or tandem connections. Furthermore, as the local network only contains a two-level exchange hierarchy and in the digital case there are fewer exchanges (tandems) on the highest level, fairly simple introduction strategies can be outlined. It also appears that there is little difference between overlaying and replacement when located exchanges are concerned.

A convenient introduction strategy could be to install a digittal tandem at an early stage and let that tandem take care of connuctions between the analogue and the digital network. Such a solution would be in favour of the final network structure and it would support the new technique and help avoid further development and extension of existing analogue exchanges. When the old parts are replaced the "transfer" capacity of the tandem exchange will be used for the growing "normal" tandem traffic. If this increases slowly, the "overcapacity" in the tandem exchange may be used for concentrators connected directly to the tandem exchange. To what extent this method is applicable in practice is of course dependent on transmission and signalling in the network, and above all it depends on cost factors. Therefor, firm conclusions could only be drawn after a thorough study of the actual case.

3.2.1. Numbering

Together with the planning of the network structure a long-term numbering plan should be made up and number series be allotted to new exchanges in accordance with that plan. In order to avoid unnecessary confusion one should adopt the principle that concentrators be included in the number series of their parent digital local exchange. As a matter of fact it is most likely that all the subscriber subsystems, remote concentrators included, that are connected to the same digital group selector will be contained in one common free number group.

Sometimes an area will have enough subscribers only for some concentrators but no for an own group selector. The concentrators will be connected to a group selector in another area until it is economical to introduce the group selector. In such a case the best solution is to include the concentrators in a number series planned for the future parent digital group selector.

3.2.2. Transmission Systems

As mentioned in chapter 2 the European PCM hierarchy will probably be 2, 8, 34, 140 and 565 Mbit/s (30, 120, 480, 1920 and 7680 channels). The 2 Mbit/s transmission is conveyed on the existing type of pair-cables, while special cables (shielded pair or coaxial cable) are required for the higher systems. Radio link transmission is not of interest in urban areas. Of special interest is the 140 Mbit/s transmission on small diameter coaxial cable.

Between analogue exchanges it is normally profitable to use PCM on routes which are longer than 6-12 km, while between analogue and digital or between digital exchanges it is profitable to use digital transmission regardless of the distance.

The digital transmission network is built up of digital sections which are common to many routes of various size and types. The capacity of the section is consequently much larger than that of the routes. In the beginning the digital sections will be combined to a star-type network, but later on a change towards a mes-htype network will occur.

The larger metropolitan areas may in the future consist of a mesh-type network of 140 Mbit/s systems between the central large exchanges and a star-type network of 2 Mbit/s systems to the periferical exchanges.

3.2.3. Transmission Planning

The reference equivalent is the most important parameter for transmission planning. CCITT recommends not only upper limits but also mean value for the national reference equivalents. The mean value of the sending reference equivalent (SRE) should be between 10 and 13 dB and the receiving reference equivalent (R-RE) should be between 2.5 and 4.5 dB. The mean overall reference equivalent fo international connections will then be 13-18 dB.

For the transmission planning in a network developing according to the overlay principle the following applies. With the introduction of PCM on the longest routes between the local exchanges, the reference equivalents for the longest connections will be reduced. The further development with digital switching and transmission will give more subscribers a lower value on SRE and RRE. When digital telephone sets are introduced, the reference equivalents for these can be chosen to a suitable value giving a more optimal value for the distribution of reference equivalents. The "fully digital goal" when all switching and transmission is dgital will result in all subscribers having identical values of SRE and RRE. The transmission planning will thus consist of among other things, planning the distribution towards an ideal value which also includes the avoidance of too low values for the reference equivalents.

The replacement of an analogue exchange with a digital one, while keeping the analogue transmission links, is equivalent to an introduction of an extra transmission link. If the analogue links are two-wire, the digittal exchange will from a small four-wire loop, where the stability has to be mentioned. One way to do this is to insert build-out pads, which results in the distribution of the reference equivalents being changed to higher values. On the other hand the transmission links will be extended by or replaced with digital links. Then the development will be similar to that sketched for the overlay principle.

3.2.4. Signalling

A digital network, overlaid or replacing the existing analogue network, should use common channel signalling (CCS).

The amount of signalling equipment for interworking with the existing analogue network is minimized if the interface between the analogue and the digital networks is not crossed more than once for a particular call. The amount of signalling equipment is also reduced if the interworking can be carried out at higher levels in the network. This means a better use of the equipment and also makes it possible to completely avoid MFC-signalling in digital exchanges at lower levels.

On a PCM-route between a digital exchange and an analogue SPC-exchange, the same CCS system as in the digital network should be used. In case of analogue transmission links, an analogue version of the CCS system can be used, or a quasi-associated mode of operation can be employed.

On a route between a digital exchange and non-SPC analogue exchange. PCM is prefered together with a proper type of channel associated signalling.

The signalling system between the digital local exchange and a concentrator is largely dependent on the switching system. It may be based on thesame principles as used in the CCS-system for autonomous exchanges.

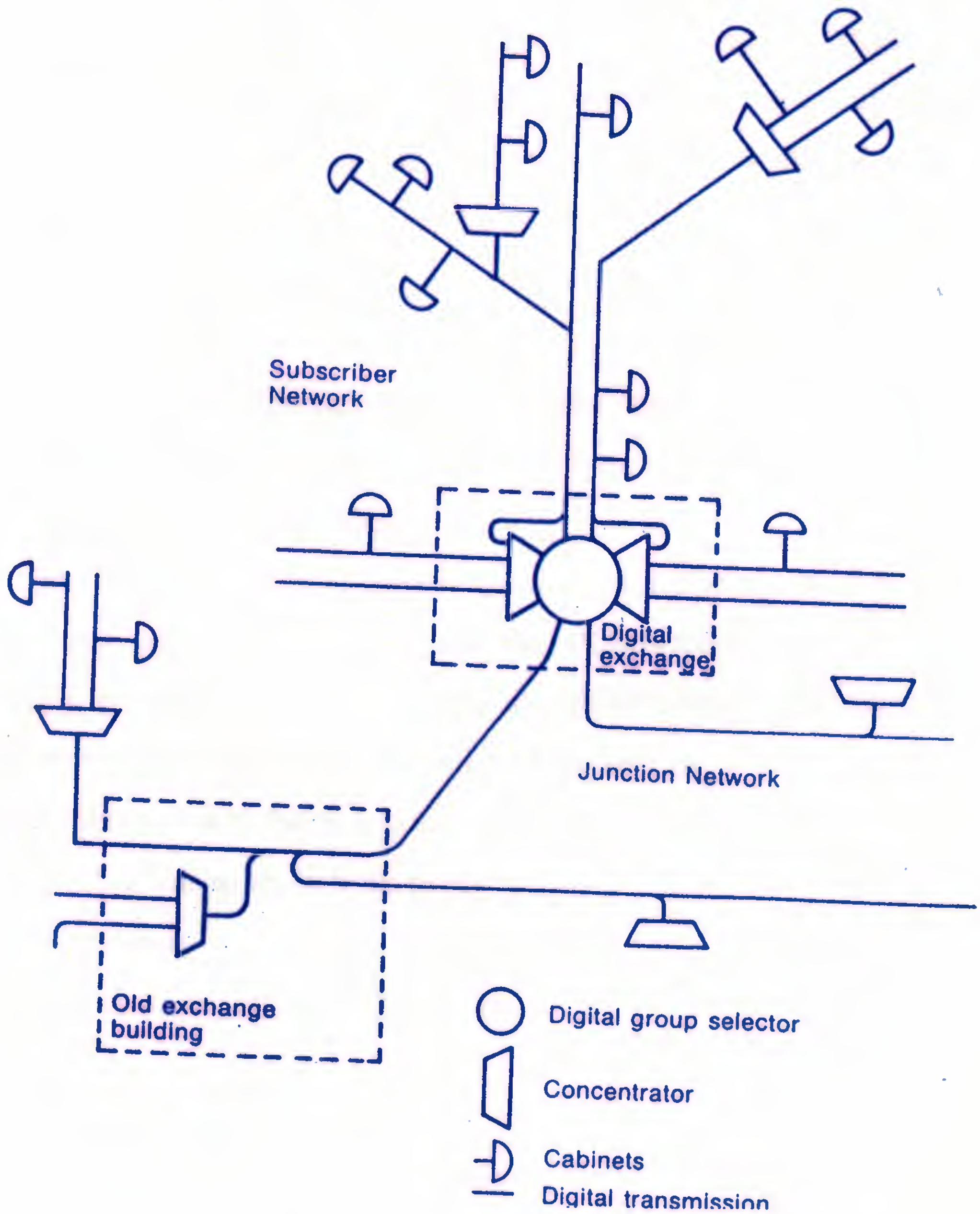

Figure 3-4 *Different locations of concentrators*

3.3. Network Structure in the Rural Area

The information presented so far has treated solutions for urban areas. There we will have large switching points and, due to the hierarchical structure and alternative routing, a high utilisation of the transmission media. For the long distance network the sittuation is similar, with a slight difference in the introduction of digital technology depending on the different transmission medias used existing SPC-exchanges etc.

An area asking for quite different solutions is the rural area. Due to large distances it has been difficult to find economically feasible solutions with existing centralized switching. This, together with other factors, has in many countries resulted in a lower telephone penetration in these areas.

The need for decentralized switching may be met by distributed concentrator systems controlled from a group centre and, if economical, with internal switching facilities.

The low utilization of transmission media due to small groups can be overcome by a common use of the PCM-system by different concentrators. All this will give us a new structure of the rural network with remote controlled concentrators linked together by a common PCM system. The network will have a good utilization of transmission media and group selector inlets as well as plossibilities of centtralization of common control functions.

Different examples of structures are given in Figure 3.5.

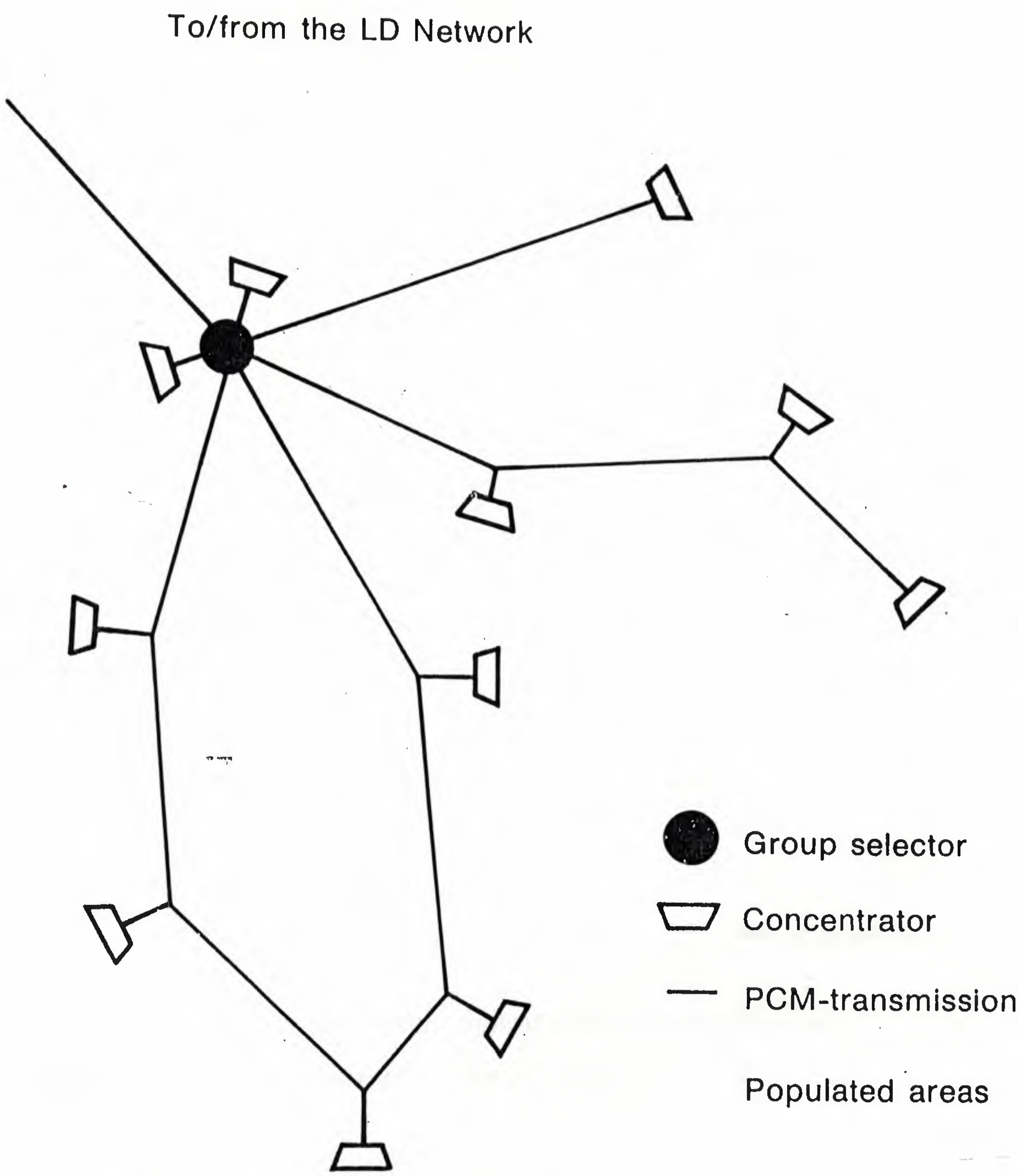

Figure 3.5. *Different structures for a rural network witth concentrators.*

3.4. A Completely Digital Network

The trends described above should ultimately lead to a completely digital network. This implies the penetration of digital transmission down to the individual subcriber set, where analogue/digital conversion is performed. As mention abov ethis penetration is a long-term one. Apart from the technical and economical problems, an international standardization is desirable before any large-scale introduction of digital telepbhone can take place. A model of a completely digital network, applied to a multi-exchange area, is shown in Figure 3-6.

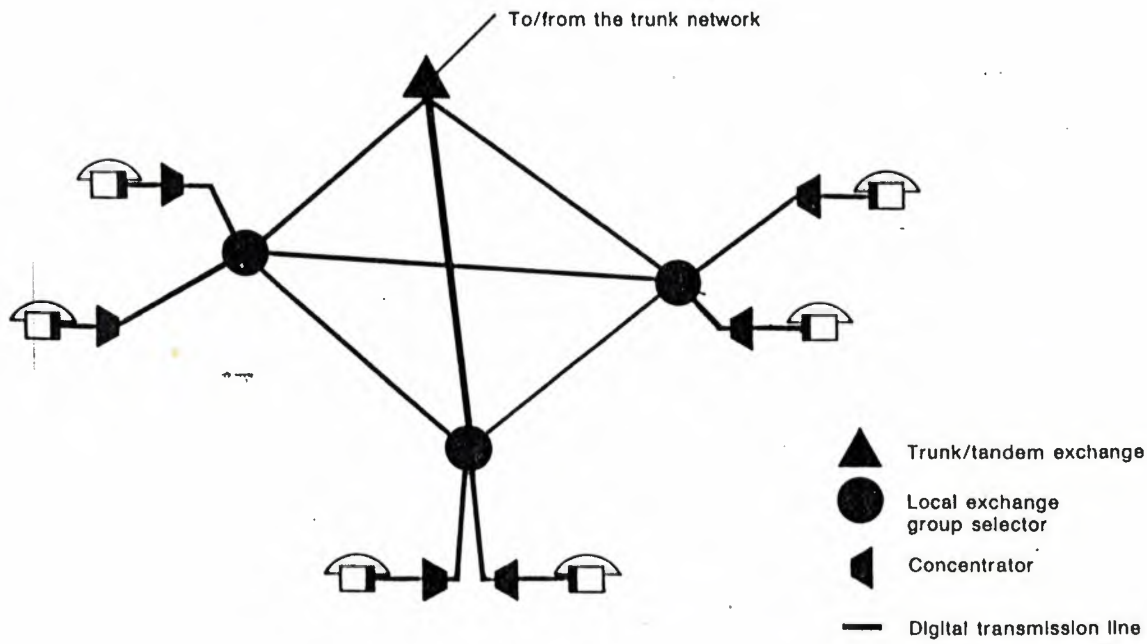

Figure 3.6. *A completely digital multiexchange network. Analogue/ digital conversion is performed in the subscriber set.*

TThe network consists of large local exchange group selectors connected in a mesh structure, controlling a number of digital concetrators, which may be remotely located or colocated with the group selectors. The concentrators together with their group selector from a star structure. Access to the trunk network is arranged by a digital trunk exchange. At this point a tandem exchange may be situated, switching overflow traffic between the local group selectors. All transmission is digital. Analogue/

digital conversion, using for example PCM or delta modulation, is performed on a per channel basis in the subscriber set.

Very much has been left out of the picture. The service integration of telephony, telex and data has not been considered, neitther has the influence caused by visual telephone, cable TV etc. TThis total picture cannot be foreseen for the present.

CONCLUSION

The project provides a strong comprehensive and illuminating a knowledgement about PCM technique and itts application in digital telephony.

The outors have tried to present in detail basic operations of PCM technique: saampling, quantizing, compaunding and coding.

The project can be used by students in the laboratory of "Telecommunication" and by engineers in design PCM system and telephone network.

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