DIGITAL TRANSMISSION OVER EXISTING ANALOGUE TELECOMMUNICATION SYSTEM OF BANGLADESH.

BY

MD. MUJIBUR RAHMAN

A THESIS

SUBMITTED TO THE DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING IN PARTIAL FULFILMENT OF THE REQUIREMENTS FOR THE DEGREE OF MASTER OF SCIENCE IN ENGINEERING (ELECTRICAL AND ELECTRONIC).

DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA, BANGLADESH.

NOVEMBER, 1987.

CERTIFICATE

THIS IS TO CERTIFY THAT THIS WORK WAS DONE BY ME AND IT HAS NOT BEEN SUBMITTED ELSEWHERE FOR THE AWARD OF ANY DEGREE OR DIPLOMA.

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Training and Research(ICTVTR), Tongi,Dhaka.

ACKNOWLEDGEMENT

The author would gratefully remember and pray for peace of the bereaved soul of Professor A.K.M. Mahfuzur Rahman Khan, Professor of the Department of Electrical and Electronic Engineering, BUET, Dhaka who supervised this work over a long period.

The author expresses his profound gratitude and indebtedness to his Supervisor Dr. Md. Abdul Matin, Associate Professor of the Department of Electrical and Electronic Engineering, Bangladesh University of Engineering and Technology (BUET), Dhaka for his sympathetic and valuable guidance throughout the work. Without his help and encouragement this work could not have been a success.

The author would express his deep sense of gratitude to Professor Md. Mujibur Rahman, Professor and Head of the Department of Electrical and Electronic Engineering, BUET for all-out support towards successful completion of the program.

The author is grateful to Professor S.U.Ahmed former Dean and to Professor S.F. Rahman Dean of the Faculty of Electrical and Electronic Engineering, BUET for inspiration and encouragement.

The author would remain thankful to Dr. K.M.Ahmed Associate Professor and to Dr. M.R.Khan Assistant Professor of the Department of Electrical and Electronic Engineering, BUET for valuable discussions and suggestions.

The author gratefully acknowledges the help extended to him by Mr. Mirza Ahmed Hossain, Director Research, Dr. Mahbub-ul-Haque, Divisional Engineer, Cable, Mr. Ashraful Alim, Divisional Engineer Planning of B.T&T Board for their kind help and valuable advice during the progress of the work.

Sincere thanks are due to Mr. M.A. Jalil for typing the manuscript.

Finally, the author wishes to express his appreciation to many friends and collegues who helped in carrying out various investigations related to this work.

ABSTRACT

Digital techniques are fast moving into the heart of telecommunication systems. A thorough study on the technical problems that may evolve with digital system working in analog environment and design of different parameters required for digital communication in our network were performed in this thesis. In this connection firstly, an extensive survey of the telecommunication system of Bangladesh was made.

Secondly, measurement of physical parameters such as traffic-flow data and cable characteristics of the important sections of the telecommunication system was taken. With the measured data analytical study of the quality of digital transmission in analog environment was made. Efforts have been made to design different parameters such as repeater spacing, number of repeaters required in each section etc. for a digital transmission over the telecommunication network of the country.

Finally, a study of the commonly available digital transmission system in the world market and investigation of the suitability of them for our present system has been carried out.

CONTENTS

CHAPTER - 1

GENERAL INTRODUCTION.

1.1 Introduction:

Telecommunication network is an essential infrastructure of a country. The importance of Telecommunication for socioeconomic development is manifold due to its diversified contribution in the process. On the one hand, this network serves its users for their smooth maintenance of daily life and on the other as one of the vital components of national infrastructure, it contributes to a large extent in the process of producing goods and services by the other Productive Sectors of the economy. Therefore, the necessity of telecommunication services for a developing country like Bangladesh needs no further elaboration.

1.2 Past of Telecommunication in Bangladesh:

Telecommunication in Bangladesh has a wide past history[1]. British rulers of India created Post, Telegraph and Telephone (PT&T)services in India, Primarily for administrative interest. Morse Telegraph Circuits connected all the Provinces of India with Delhi. District towns and Sub-divisional towns were also covered. Physical lines were constructed for telegraph circuits. During the end period of British rule magneto and CB-exchanges were also installed in important cities. Calcutta - Assam and Calcutta - Burma trunk lines passed across Bangladesh with repeater amplifiers. River crossing at certain places were done by under water cables.

After creation of Pakistan, telecommunication got importance because of geographical separation of two wings. So overseas H.F. Radio communication was established between East and West Pakistan. H. F circuits with London, Bern, and other important Western and Eastern capitals were installed. At the same time VHF Radio Relay lines were established in 1954-1958 to interconnect the district towns. 12 channel land line carriers were also established for connecting many sub - divisional towns.

Capital city of Dhaka inherited a 300 line CB exchange from British India and it grew upto 900 lines. In 1947 the whole country had 1500 line units of manual telephones. In 1954 an automatic exchange of 4000 line unit was installed at Dhaka. It was a Step-by-step Strawger type electromechanical switch from Siemens of West Germany.

In colaboration with Siemens a factory for EMD equipment was established at Tongi, Dhaka in 1968. A Telephone cable industry was also established at Khulna in the same year. Since then rapid automisation of switches took place.

For improving inland and overseas transmission 960 channel broadband Microwave (MW) Radio Relay transmission network and a Satellite Earth Station were planned in 1966-68. Dhaka-Chittagong and Dhaka - Khulna MW links were constructed in 1970, and came into operation in 1972. Similarly the Earth Station at Betbunia came into operation in 1975. Then H.F. circuits were gradually closed down. Dhaka-Sylhet, Dhaka-Bogra-Rangpur-Dinajpur and Dhaka-Kushtia-Chuadanga to Calcutta MW links were established by 1978. By now all the old district towns and some old Sub-divisional towns were connected by U.H.F.Sparlinks with the MW back-bone trunk circuits.

1.3 Research Activities on Telecommunication System of Bangladesh:

The present state of art of the global telecommunication is the fruit of researches by scientists and engineers for the last fifty years. In Bangladesh telecommunication is entirely a government enterprise. Extensive research works done on this system is, however, very limited. Rouf[2] studied the telegraph traffic of Dhaka-London route along with the technical losses, hourly booked and effective calls on Dhaka-Karachi route and subscribers trunk dialling in Dhaka-Chittagong route. Das[3] worked on noise in microwave communication in Bangladesh. Khan[4] studied noise and interference in some sections of the telecommunication system. Alim[5] made a statistical analysis of the overflow traffic in telephone networks. A simulation technique of telephone traffic flow model in a computer was developed by Khan[6]. Haque[7] proposed some algebraic coding schemes for transmission of Bengali speech signals. Rahman[8] worked on alternate routing strategy to enhance call handling capacity of a multiexchange system. Alam[9] studied the effect of nation-wide dialling on the telecommunication network of the country. Power line carrier (P-L-C) telephony is usually used by Bangladesh Power Development Board. An investigation into the causes of signal to noise ratio reduction in a P-L-C system has been carried out by Alam[10]. Microprocessor controlled exchange is an achievement of modern technology. A study in this line was carried out by Majumder[11].

1.4 Present State of Art of the Project:

Our country has been putting its best possible efforts[12] in order to develop a national telecommunication network and ensure its efficient operations since its very birth. Bangladesh has adopted development plans, applied relevant technolgies,, organised institutions and has been undertaking all the relevant efforts with a view to improving the telecommunication services. Mention may also be made that the country's efforts to that direction is not only confined within developing the urban telecommunication network, relying on the revenue earnings, but also extended to rural areas with a view to bringing the vast majority of the population within the national telecommunication network. Inspite of the above, telephone density at present in Bangladesh is one of the lowest in the world. The percentage of rural telephones is approximately slightly above one tenth of the total strength.

Bangladesh telephone network[13] entered into the digital era in late 1983 straight from step-by-step technology with the commissioning of four digital trunk (transit) auto exchanges at Dhaka, Chittagong, Khulna and Bogra and one digital international trunk exchange at Dhaka. Venturing into this latest technology was made possible through intensive studies on related fields on bilateral and International Telecommunication Union (ITU) assistance.

In Bangladesh Digital transmission systems were finding application in some part of the junction network even in the analogue environment, as it provided significant economic advantages over certain distances. Four PCM systems (4 x 30 chl = 120 channel) between Moghbazar and Narayanganj and Two PCM systems (2 x 30 = 60 chl) between Moghbazar and Mirpur have already been installed[14]. These are 2 M bit per Sec. 30 channel PCM systems.

In Bangladesh, cables required for the subscriber lines and analog junctions and trunk networks are obtained from the national factory, namely Bangladesh Cable Shilpa Ltd. High quality cables of international standard are manufactured here. All these cables are of symmetrical type.

For introducing PCM in large scale in the existing analogue system a careful study should be made. For this purpose a strategy for digitallization of the existing analogue system should be formulated. One redeeming feature of the networks in Bangladesh is that it is small and once the process of digitalisation starts the digital systems in the network will very soon outnumber the existing systems and the benefits of integrated digital switching and transmission may become perceptable in a shorter time frame. The planning activities i.e. forecast of requirement of circuit blocks, studies on developments in digital transmission, routing plan for automisation of trunk traffic, signalling plan, transmission plan and synchronisation plan, formulation of technical specification for the system, should be followed by the preparation of a master plan. Obviously digitalisation is possible only in discrete steps. In this context the various options available to decide on the strategy to be adopted for given situation become quite relevant. In case of Bangladesh Telecommunication these aspects should be considered carefully.

Various options available[15] are enumerated to emphasize the impact on the planning efforts.

(i) All new equipments may be of digital type, the digital systems appearing in a stand-alone manner at various parts of the network directly interworking with existing network, A/D conversion being an essential part.

(ii) Digital pockets in network may be established, Other parts in the network still growing with existing type.

(iii) Overlay approach where the digital systems and existing systems shall work side by side as separate network,access between each other being provided through tandems which limits the A/D conversion as well as signalling interworking points.

(iv) Total replacement of existing network by a digital network which can have relevance in portions of the network catering to a small area.

As it is well assumed, for efficient, economic and reliable operation of any communication system, the digitalisation on existing analogue environment needs a careful and thorough study. The physical prameters of the cables especially the insulation resistance, loop resistance, capacitance and attenuation of the existing cables will be measured and analytical study will be made to recommend a PCM system that can be economically employed in Bangladesh Telecommunication network.

1.5 Objective of the Research:

Digital techniques are fast moving into the heart of telecommunication systems. In Bangladesh, this has already been started in the nationwide dialling system and in some isolated PCM links. Before going into large scale digitalization of the system, a thorough study on the te- technical problems that may evolve with digital system working in analogue environment all around should be made. With the available resources in Bangladesh university of Engineering and Technology (BUET) and Bangladesh T&T Board, measurement and study of the system characteristics can be performed successfully.

The objectives of this research are as follows:

(i) To make an indepth study of the telecommunication transmission systems of Bangladesh in the context of the planned introduction of digital communication facilities.

(ii) To identify the main features of digital systems that may be served best by the analogue systems available.

(iii) To identify the critical factors that will determine the reliability and successful working of the digital system.

(iv) To study the workability of major digital systems in this analogue environment.

(v) To investigate the problem of compatibility and interfacing of possible digital systems with the analogue systems present.

For insight into the existing analogue telecommunication environment an extensive survey of the present communication networks of the country has been performed in chapter-2. The parameters of the existing system such as traffic overflow, call handling capacity, problems with multiplexing of analogue signals etc. and existing cable characteristics were studied in chapter-3. Chapter-4 presents systematic methods of planning and design of digital communication network using the existing cables. In chapter-5 a comparative study of the characteristics of commercially available digital systems was performed. The thesis is concluded with a general discussion in chapter-6.

CHAPTER-2

AN EXTENSIVE SURVEY OF THE TELECOMMUNICATION SYSTEM OF BANGLADESH

2.1 Introduction:

Bangladesh is a subtropical country covering world's largest delta zone where rivers occupy nearly 10% of the whole territory. The most part of the territory consists of flat land and during the monsoon these flat-land regions submerge. The typical climate is featured in much rains and high humidity. Bangladesh has poor railway, road and other means of physical communication. During and after rainy season many places are isolated by flood and telecommunication provides the only means of quick contact. A survey of the existing system is given below.

2.2 Existing Telephone System of Bangladesh:

At present Bangladesh has 1,62,000 automatic and 20,000 manual telephone installed in its network.[1] Thus total installed capacity is 1,82,000. This gives a density of 1.8 telephone per 1,000 population. However, rural penetration is very low. In the upazillas 144 magneto manual exchanges are installed. But almost all the District towns have automatic exchanges. In all Bangladesh T&T Board has 80 automatic exchanges in its network. Almost all the local auto exchanges work with step by step equipments either FI or EMD Siemens type.

For nation-wide dialling (NWD)[16] 4 digital electronic trunk exchanges have been installed at Dhaka, Chittagong, Khulna and Bogra. So far 25 districts have been integrated with the nationwide Dialling(NWD)network. The basic configuration of the network was planned to be radial superimposed by a meshed network with direct routes working as high usage group with alternate routing and overflow from direct routes to other direct routes and finally to last choice routes. The basic network will ultimately consist of 4(four) network levels (Fig. 2.1).

- Main exchange area(ME) i.e. national transit switching centre.
- District exchange area (DE)
- Sub district exchange area (SE)
- Terminal exchange(TE) or local exchange area

Numbering Plan:

The numbering plan is based on the final construction stage with sufficient reserve for 50 (fifty) years planning period according to CCITT recommendations. It has been taken care that the area codes for local networks with high interest of communication will be as short as possible.

Access to NWD network is provided by dialling the traffic discriminating digit '0' whereas that for international Trunk exchange (ITX) is '00'. Trunk codes are normally 3 digit ABC. But for some stations it is of 1 and 2 digits excluding '0' e.g. area code for Local network 'Dhaka' is '2' and that for Chittagong is '31'. Calls to the centralised special services type IX(X) and IY (X) Q for nationwide dialling (NWD) network are routed to the sub-district exchanges, main exchange or in exceptional cases to the International Exchange.

For example: Type 1X(X) : 150 ... AAT (Auto answer trunk)

151 ... TAX ASC Booking.

152 ... 6 ITX ASC Booking.

Type IY(X) : 160 ... STC (System Test Console).

161 ... TAX ASC enquiry.

162 ... 6 ITX ASC enquiry.

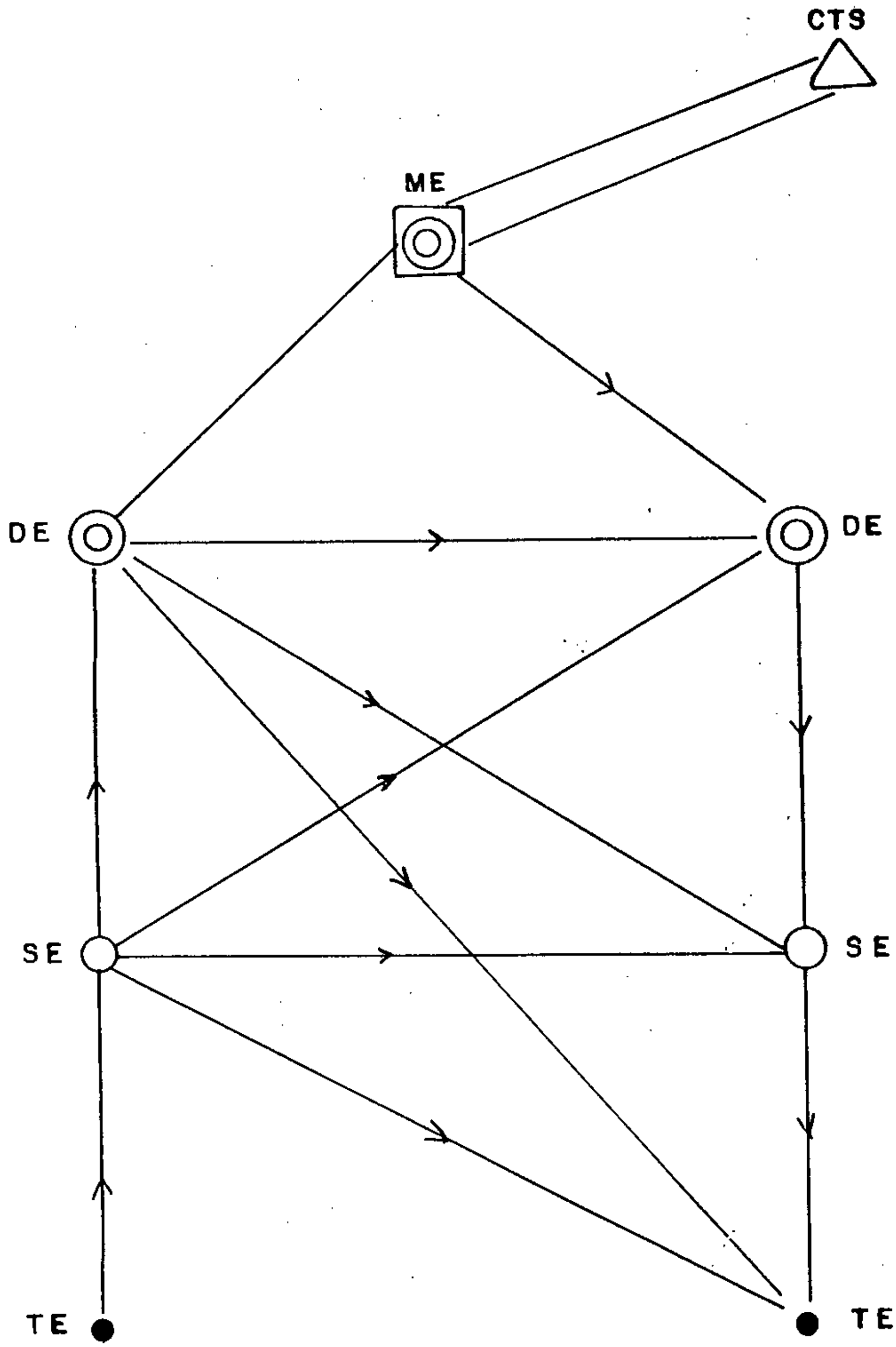


Fig.2.1 Telephone network level of Bangladesh

In case of congestion, failure or non-attendence of a particular ITX ASC service call will be automatically diverted to any one of the free ASC. The same ASC position can be used for booking, enquiry and for termination of calls. The country code of Bangladesh is 880. The national number can obtain maximum 8 digits and minimum 6 digits without prefix '0'.

Installed capacity of trunk automatic exchange (TAX).

Installed capacities of trunk automatic exchanges are as follows:

Name of TAX.	Incoming Trunk.	Outgoing Trunk	Total Trunk
Dhaka	2064	2007	4071
Chittagong	760	720	1480
Khulna	573	546	1119
Bogra	587	606	1193

2.2.1 Local Network:

A Local network[17] is the outside plant for junction and subscriber circuits (Fig. 2.2). A local network may comprise more than one exchange (each having its own area and network) and include those sections of junctions, cables etc. which lie within boundaries of local area. Local network may consist of two : namely,

(a) Subscribers line network that connects the subscriber to its telephone exchange.

(b) Junction network that connects the exchanges in a multi-exchange area.

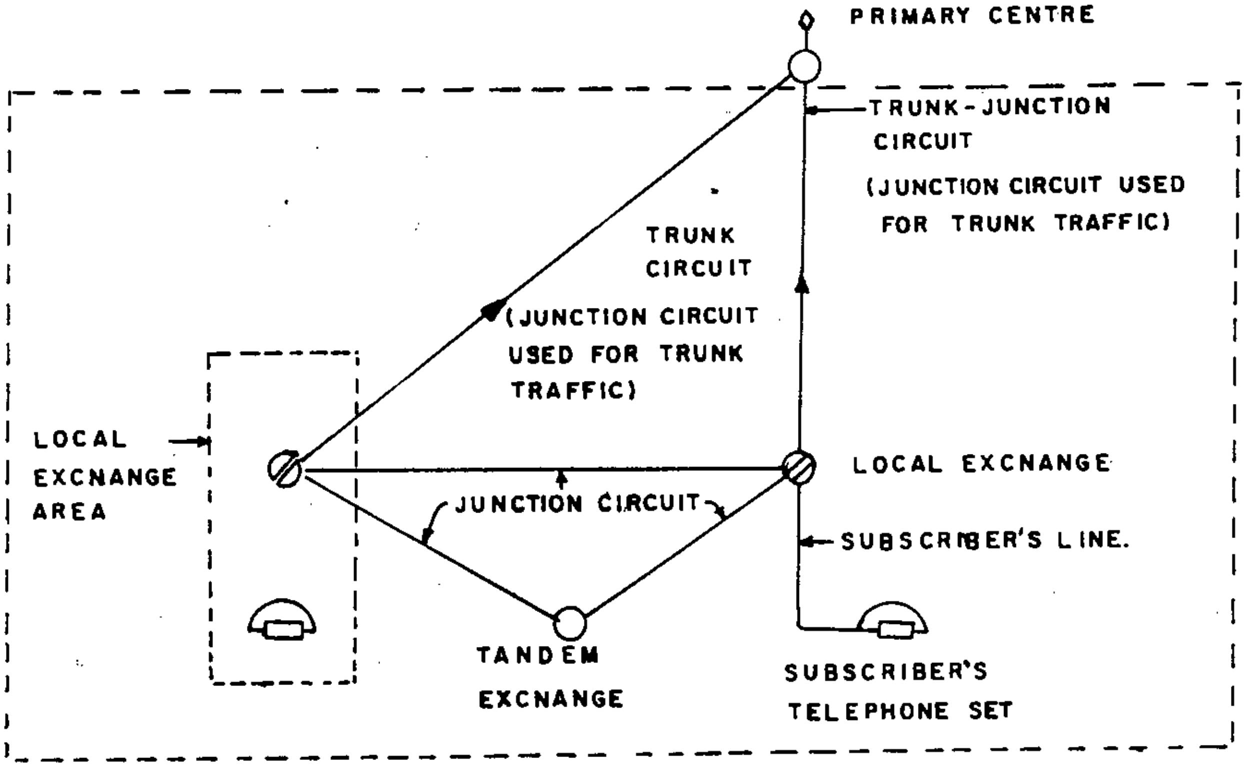


Fig.2.2 LOCAL NETWORK AREA

The local network which originate and terminate all calls whether local, long distance or international can be termed as artery of telecommunication system[18]. In other words, the performance of a local network is a measure of the standard of a telecommunication system. According to ITU/CCITT Survey, an average of 40-50% of the total telecommunication investments on national telephone system has to be spent for the local networks and the subscriber plants.

The tables below show the local network condition of the Dhaka multi-exchange area.

(a) Primary cable : (Moghbazar, Mirpur & Gulshan exchange only)

Installed capacity: 32,950 pairs.

Working pairs : 18,541 pairs.

(b) Junction Network :

At present there exist three junction networks in three cities in Bangladesh[19]. The cities are Dhaka, Chittagong, and Khulna. The junction network of Dhaka is the largest. Obviously performance grade of the network of Dhaka largely determine the telecommunication system efficiency of the capital.

Dhaka multi-exchange area consists of eight telephone exchanges connected by junction cables. The detail of the junction cable network of Dhaka multi-exchange area is given in Fig. 2.3.

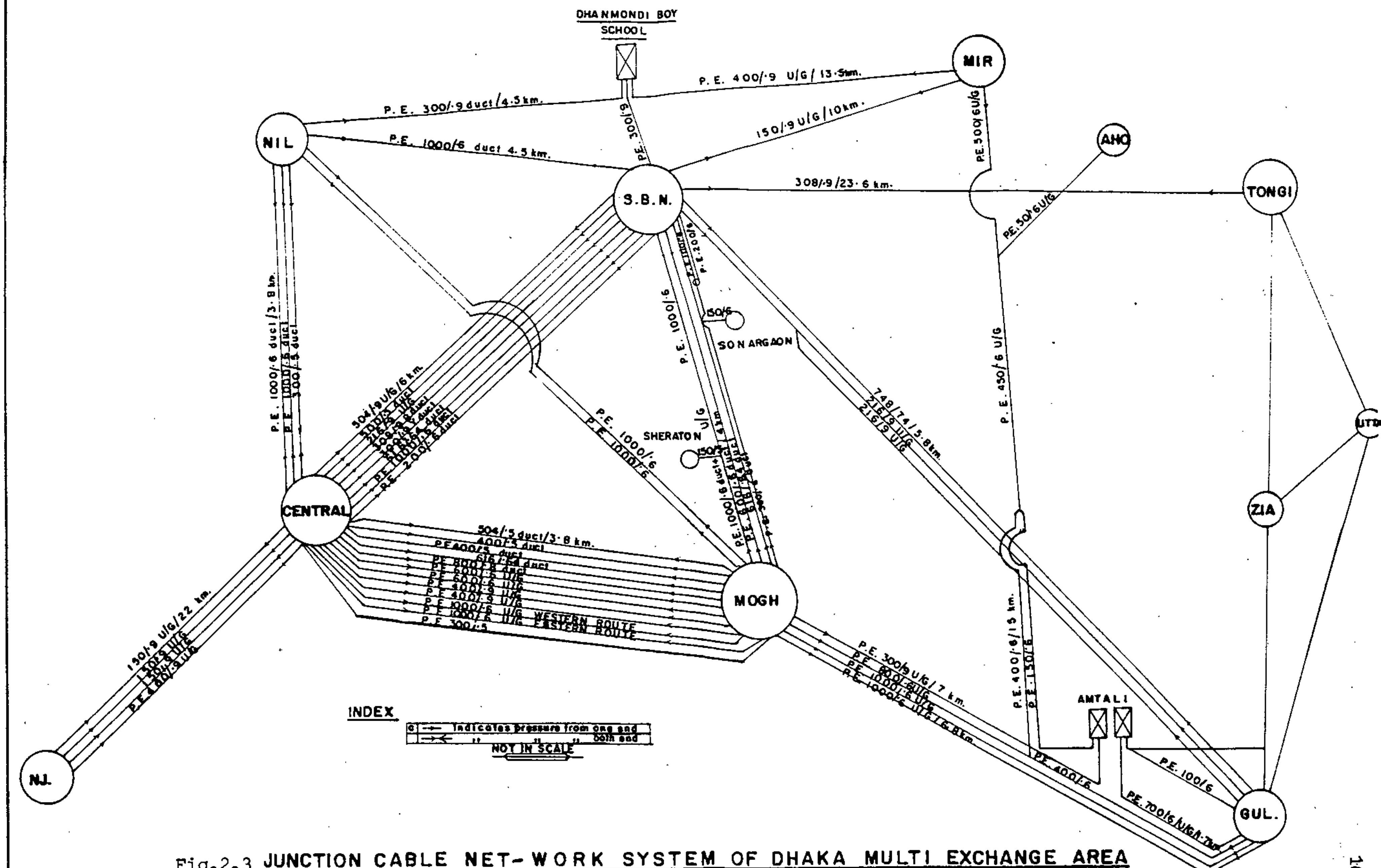


Fig.2.3 JUNCTION CABLE NET-WORK SYSTEM OF DHAKA MULTI EXCHANGE AREA

Drawn by	Prepared by	Checked by	Appr. by
D.MAN	S.A.E.	S.D.O.CABLE	D.E.CABLE

BANGLADESH TELEGRAPH & TELEPHONE BOARD

2.2.2 National network:

Bangladesh T&T Board opened its 960 channels[20] of broad-band Microwave links in 1972. At first Dhaka-Chittagong and Dhaka-Khulna MW links were installed. Then Dhaka-Sylhet, Dhaka-Mymensingh and northern lines were constructed. Dhaka-Khulna link was extended upto Barisal and Dhaka-Chittagong line was extended upto Cox's Bazar. 25 Hops of broad band MW-networks are established from Atwari in the north of Dinajpur to Teknuf in the south, Sylhet in the north-east, Barisal in the south and Chuadanga in the west. 25 UHF sparlinks are connecting rest of the district towns with the MW backbone. In Bangladesh MW links covered 1200 km distance. The total channel capacity is about 15,000.

These link, formed the back-bone trunk routes for nationwide telecommunications. All these links are of analog technique based on FDM-FM system. The main links have a capacity of 960 channels.
The following brands are connected to the links:

LENKURT	(UPPER 6 GHz band) for	DHAKA-CHITTAGONG
		DHAKA-KHULNA-MONGLA.
FUJITSU	(UPPER 6 GHz band) for	DHAKA-BOGRA-ATWARI
	(6 GHz band) for	PHULBARI-RANGPUR
		DHAKA-MAGURA-KUSHTIA(and to India).
		BOGRA-KUSHTIA.
		NATORE-RAJSHAHI
		TANGAIL-MYMENSINGH.
TOSHIBA	(6 GHz band) for	DHAKA-SYLHET

RCA (6 GHz band) for CHITTAGONG-COX'S BAZAR.

(2 GHz band) for CHITTAGONG-RANGAMATI

This main network is star shaped around Dhaka and Provides links from Dhaka in the most important directions:

South - East	(to Chittagong)
South	(to Khulna)
North - West	(to Bogra)
North - East	(to Sylhet)

The only link permitting a diversification of routes is in the western part of the country: Magura-Kushtia-Bogra. The link between Dhaka and Talibabad has a special function. This link has been provided by THOMSON-CSF, with a stand-by on Co-axial cable to connect the earth station of Talibabad to Dhaka. In addition to that an 1,800-Channel, 6 GHz (upper) of NEC analog MW transmission link has recently been installed between Dhaka and Chittagong. The links are shown in Fig. 2.4.

2.2.3 International Network :

A digital international trunk exchange (ITX) has been installed at Dhaka for overseas direct dialling and operator assisted trunk services. Powerful computers records the international and also the nationwide dialling (NWD) calls. Installed capacities of International trunk exchange are as follows:

- International Trunk (Bothway) = 319
- International Trunk (ring down) = 2

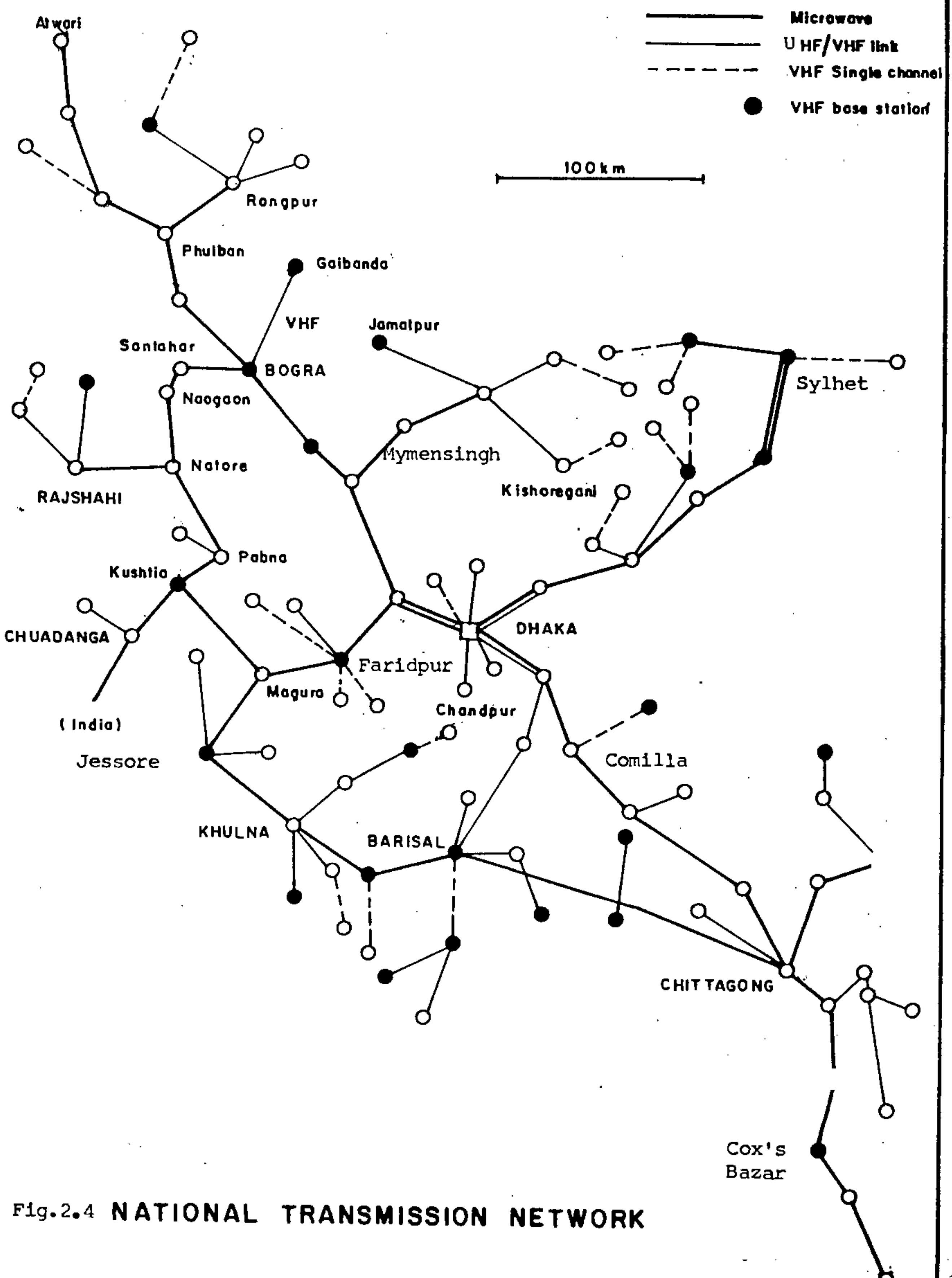


Fig.2.4 **NATIONAL TRANSMISSION NETWORK**

–	International Trunk exchange to Dhaka Trunk automatic exchange	= 117
–	Dhaka Trunk automatic exchange to International Trunk exchange	= 73
	Total	= 511

One Standard – A earth station at Betbunia and a Standard – B earth station at Talibabad are providing 130 overseas Telephone circuits and 96 Telegraph/Telex circuits with 18 overseas countries. Block diagram shows how international network is connected to our Network (Fig. 2.5).

2.3 Telegraph and Telex Services:

There are about 600 Telegraph offices in the country. Important district towns are interconnected by point to point Teleprinter circuits. In other places Morse code or Phonogram service is available. There are 5 Telex exchanges in Bangladesh. Out of them 3 telex exchanges are installed in Dhaka, One in Chittagong and one in Khulna. Besides them one 25-lines Gentex are working in Dhaka. The following are the installed capacity of Telex exchanges.

Dhaka : a) 1000 lines digital Telex exchange : NEC Japan.

b) 100 lines Telex exchange – Siemens, W.Germany.

b) 50 lines Telex exchange – Siemens, W.Germany.

Chittagong: a) 260 lines Telex exchange – Siemens, W.Germany.

Khulna : a) 25 lines Telex exchange – Siemens, W.Germany.

Chittagong and Khulna are connected with Dhaka Telex by line concentrators.

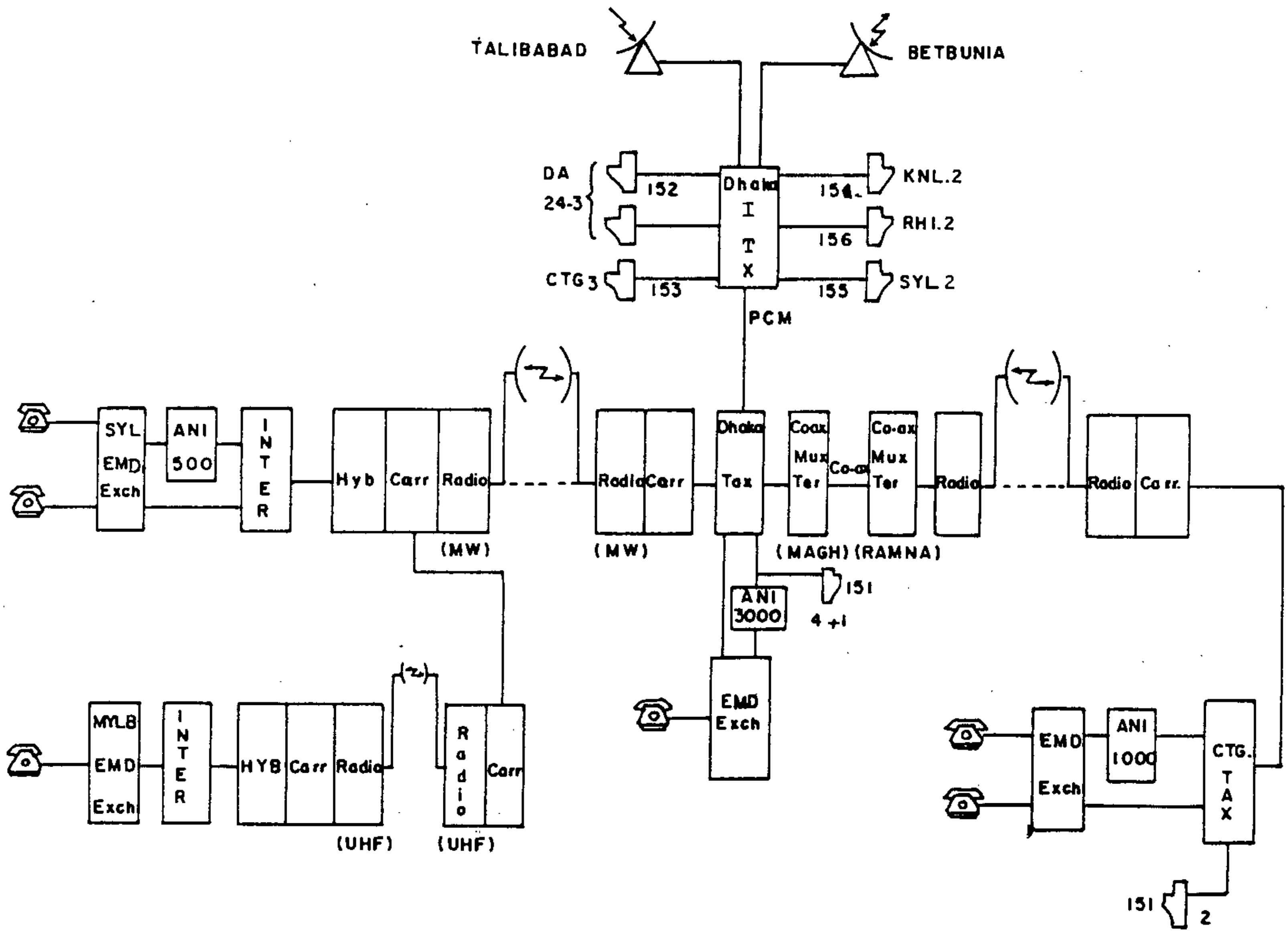


Fig.2.5 International network connected to Bangladesh network

2.4 Discussion:

The population density of Bangladesh is one of the highest in the world. For 100 million people of the country has about 0.18 million telephones only which is one of the lowest in the world i.e. 1.8 telephones per 1,000 population.

Telecommunication plays a vital role in improving infrastructure for economic development of a country. For Bangladesh this is more important because of her poor development in other sectors of communication. Although pace of development of telecommunication was rapid in the past decade, it could not be made faster with the age old system existing in the country which have already been discarded in the developed countries of the world.

With a view to expand and modernise telecom. networks in Bangladesh, a careful decision should be made so that developments in the relevant field becomes appropriate and most economical. The present research aims at this goal in respect of digital transmission over existing analogue environment and to suggest a national digital transmission system which will be suitable in our network.

CHAPTER-3

PHYSICAL PARAMETERS OF THE EXISTING ANALOGUE NETWORK.

3.1 Introduction

For assessment of the improvement expected from the introduction of digital transmission over the existing analogue signalling system it is necessary to study the physical parameters of the existing system. The essential parameters for this purpose are the traffic overflow, call handling capacity, the problems associated with the multiplexing of analogue signals and the characteristics of existing cables. On the basis of these parameters we can judge the quality and extent of application of digital transmission systems.

3.2 Traffic Overflow

Study of overflow traffic is important for efficient planning and management of telecommunication systems. Trunks are most efficiently used when properly loaded but blocking is increased with rise of traffic.[5] Overflow systems are economic solutions to this contrary demand. Overflow system helps the practice of concentrating traffic from less costly lines to more costly trunks. Within the limits of a fixed grade of service, concentration of traffic from two or more groups of sources can be realized with fewer equipment if they are made to share a common secondary group in addition to their primary or direct trunks instead of feeding individual groups of trunks. Traffic to the common group in such an arrangement are those overflowed from the different primary groups, gradings, peak and top load finders. Alternate routing arrangement are common examples of overflow systems in telephone connection networks.

In our country the telephone subscribers are considered in three groups:

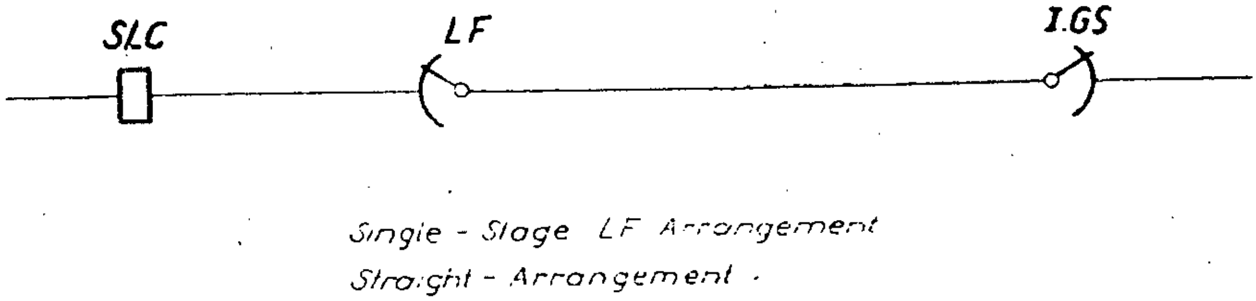
i) Normal group

ii) Heavy group

iii) PABX or PBX group.

Normal group covers the residential part of the total subscriber while the heavy group is considered to be the commercial enterprises. PABX is used in office and are always fed with maximum load. Considering the traffic pattern of each group overflow facilities are incorporated. PABX does not employ any provision for overflow traffic. Hence its switching equipments are designed with a consideration of maximum offered traffic. In the case of the arrangement of subscriber line groups in the preselection stage,in such a way that switches are well-utilised in the group selection stage.

In the EMD exchanges the LF's (Line finder) as a rule are either grouped in a straight forward trunking arrangement or in an overflow arrangement. In the straight forward trunking arrangement each line finder is permanently fixed to a Ist group selector. But in the overflow arrangement a 2-stage line finder (Fig. 3.1) is utilised which is known as LF/TLF(Top Lood finder) arrangement. In this arrangement only the LF's connected to one Ist GS (group selector) are seized first in each subscriber groups and are highly utilised. The line finders which carry only peak traffic are designated as top line finder and are connected to the free inlets of the neighbouring line groups. This causes an equalisation of traffic within several subscriber line groups.



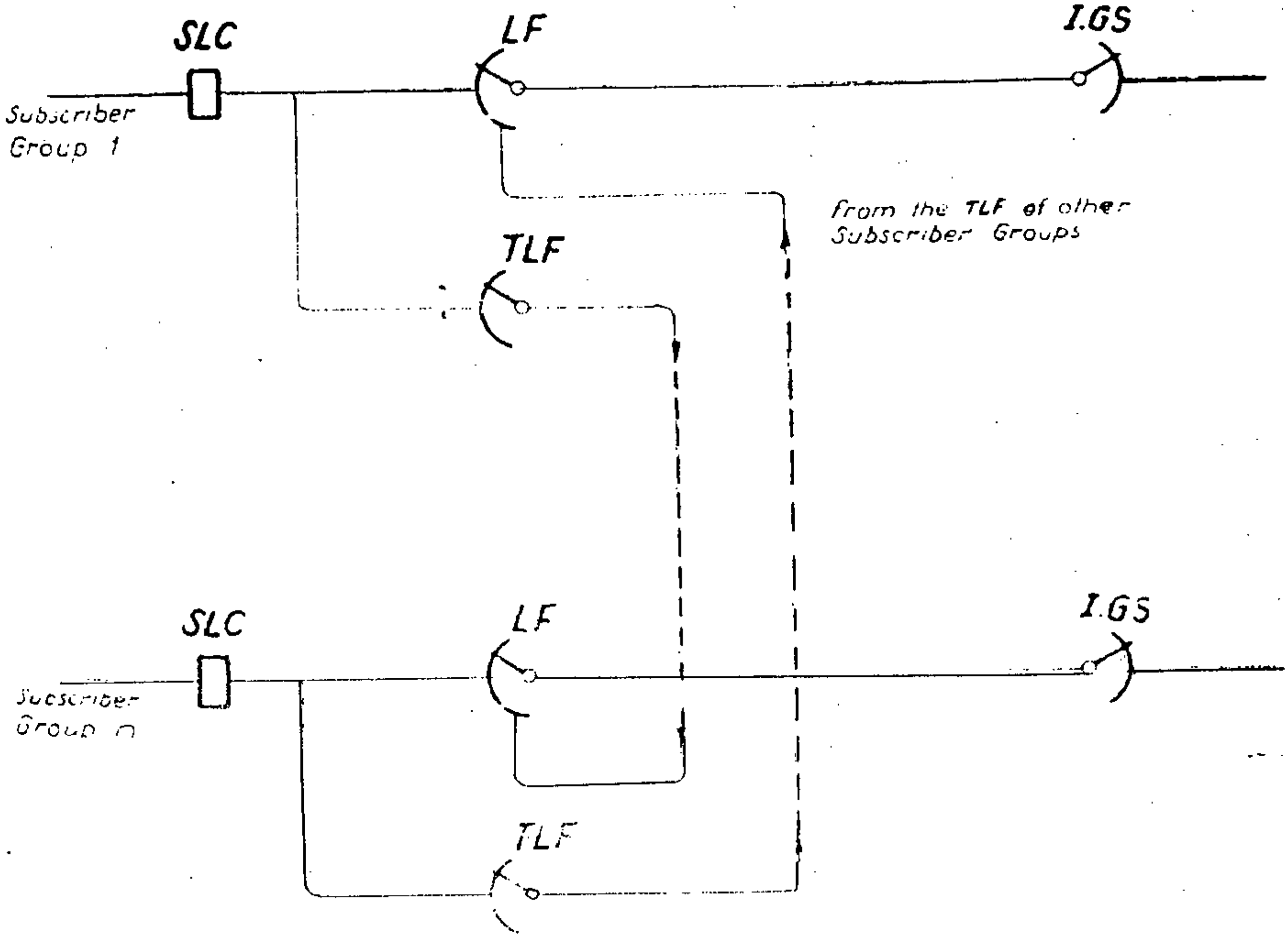


Fig.3.1 Two stage LF-arrangement with overflow for n subscriber groups

In chapter two it is mentioned that Bangladesh telecommunication system consists of mainly G1, EMD and magneto-manual exchanges. Recently Bangladesh entered into a new era of modern telecommunication through the introduction of nation wide dialling (NWD) system. So the analysis of overflow/call failure should be made so that maximum utilisation of these exchanges can be achieved and appropriate trunk lines can be provided.

Before analysis of this system it is necessary to know how the terminal (peripheral) stations are connected and system performance criterion should be made. The Four Trunk Automatic Exchanges (TAX) are interconnected directly through microwave/VHF Line for convenience. The Four TAX's are interconnected as shown in Fig. 3.2. So, a Subscriber of any peripheral station can dial directly to a subscriber of another terminal station which is even terminated at another TAX. At present 31 stations are connected with 4 TAX's.

To analyse the data the following informations should be made clear.

1. Trunk hunt OK - whenever 'O' is dialled the subscribers enter into TAX through the Ist GS of Local exchange and Interface installed at the terminal station in an Interface station and directly through the Ist GS of Local exchange where the exchanges are directly connected to TAX. As for example, the subscribers of Faridpur enter into Dhaka TAX through interface but the subscribers of Dhaka enter Dhaka TAX directly. Trunk hunt OK means trunk hunting from TAX onwards Local exchange or Long distance terminal TAX or interface.

2. Trunk hunt NG-Trunk hunt'Not Good' means non-availability of aforesaid trunk as in 1.

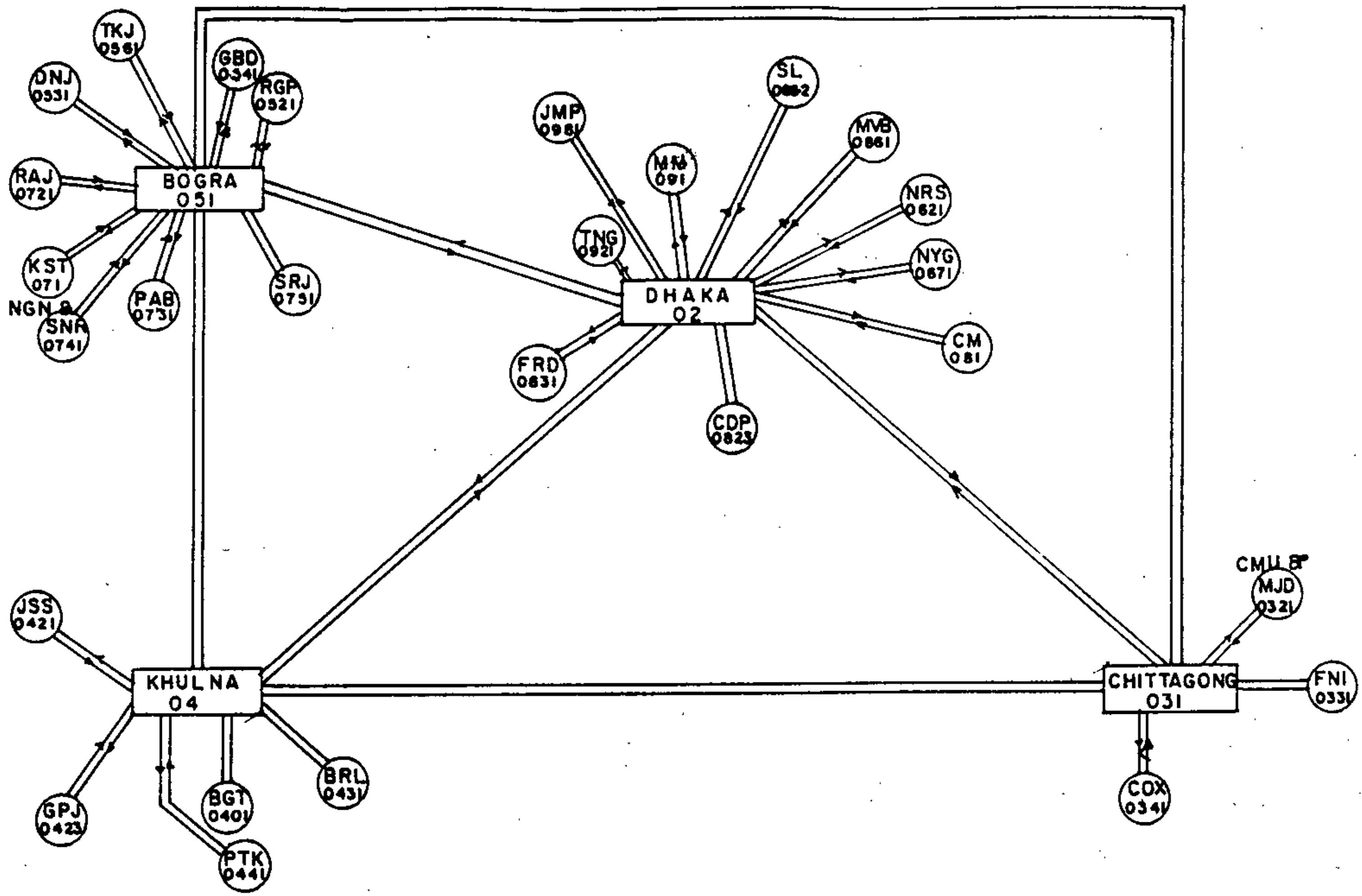


Fig. 3·2 Topological Diagram of NWD
Network in Bangladesh

3. Fundamental call attempts— This is the total number of calls attempted to TAX. This includes direct calls (attempted by the subscribers) and semi-automatic calls (connected by operators).

4. Incoming call attempts—Data measured as incoming call attempts are as follows:

(a) Clear just after seizure——This means dialling is abundondd just after entering TAX. In this case the subscriber did not dial completely but dialled only the digit '0'.

(b) Time out just after seizure— The inter-arrival time between two successive digits should be 10-20 Secs. If this is not maintained, the TAX will send reorder tone and the call will be cancelled by TAX.

(c) Wrong dialling pulse received by dial pulse incoming resister (DPIR)— This occurs if TAX receives wrong dialling pulse from the Local exchange. For example TAX receives 11 pulses instead of 10 pulses when '0' is dialled on any occassion.

(d) Wrong dialling pulse received by Multi-Frequency converting Resistor(MFCR)——This happens if TAX receives wrong pattern in MFC (Multi frequency code) signalling system. For example, there should be 2/6 pattern (two 1's and four '0's) in the MFC signalling code.

(e) Vacant code dialling—Subscribers dial wrong area code which is not available at TAX memory. In this case the subscribers will get announcement. For example some body dials 0631-2XXX: here 2 is not available at Faridpur at this moment.

(f) Subscribers abandon in the middle of dialling—This means the subscribers did not complete dialling. They abandoned dialling in the midway of the total digits to be dialled.

(g) Interdigital pause—This means the interarrival time between two successive digit has not been maintained. Interdigital pause is 10-20 Seconds measured at Dhaka TAX.

Traffic data on 24 hour basis of Dhaka TAX were taken for 3 days on 12, 13, 14th of May'85. The Data are analysed to check for the missed calls. An example at 11 hrs on 14th May'85 is shown below :

1. Trunk hunt not good = 9362
2. All trunk busy = included in 1.
3. Failure with incoming call attempts=
    (a) Clear just after seizure = 505
    (b) Time out after seizure = 26
    (c) called subscriber busy = 00
    (d) Wrong dialling pulse received by DPIR = 80
    (e) Wrong dialling pulse received by MFCR = 177
    (f) Vacant code dial = 5,657
    (g) Subscriber abandoned during dialling = 25,205
    (h) Inter digit pause more than specified. = 2,253

    = 31,903
4. Sender Time out = 1,845
5. DPIR Trunk hunt not good = 3,620

Summation of (1) to (5) = 46,730

6. Number of incoming call attempts = 69,061

7. No. of incoming calls that should mature=69,061-46,730

= 22,331.

8. No. of calls matured = 17,873

9. No. of call failures (7) minus(8) = 22,331-17,873

= 4,458

10. No answers back (assuming 5%) = 22,331x5%=1,117.

11. Calls that can not be accounted for

except miscelleneous reasons = 3,341.

The percentage of different types of fault and matured calls is shown in Table 3.1 and on the basis of Table 3.1 a curve is drawn and it is shown in Fig. 3.3.

From Fig. 3.3 it is evident that the rates of Trunk hunt not good and all Trunk Busy (A.T.B.) increase by (y_2-y_1) 14% and matured call decreases by (x_1-x_2) 14.8% around 12.00 hours. Thus keeping all others constant, amount of matured call depends apparently on getting of the trunk. In Table 3.2 and in Fig. 3.4-3.7 hourly offered and successful calls are recorded from Dhaka TAX. Call-data of Dhaka TAX were studied also by Alam[9]. Those data recorded in different dates also indicate poor rate of matured calls during busy hours. In fact, a massive digital transmission scheme that ensures greater call handling capacity could solve this problem.

TABLE 3.1 : Traffic Data in Percentage of Dhaka TAX

Hours	00hr	01hr	02hr	03hr	04hr	05hr	06hr	07hr	08hr	09hr	10hr	11hr	12hr	13hr	14hr	15hr	16hr	17hr	18hr	19hr	20hr	21hr	22hr	23hr
1.Trunk hunt NG	5.39	0.41		3.5	0.	0.0	0.0	1.79	6.55	8.30	12.34	13.44	15.60	16.50	16.02	15.95	13.69		12.96	13.08	12.87	12.82	11.24	11.2
2.All trunk busy					Included in 1																			
3(a)Clears just after seizure	4.33	3.88		13.7	4.57	3.56	2.95	5.71	1.87	0.78	0.63	0.85	1.13	0.27	0.86	0.70	1.07		0.7	1.25	0.83	0.85	0.67	
(b)Time out just after Seizure.	0.05	0.0	Not Recorded	0.11	0.0	0.16	0.40	0.14	0.02	0.19	0.01	0.01	0.01	0.01	0.0	0.0	0.08	Not Recorded	0.03	0.03	0.01	0.04	0.02	Not Recorded
(c)Called subscriber busy.	0	0.0		0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0		0.0	0.0	0.08	0.0	0.0	
(d)Wrong dialling pulse Rec.by DPIR.	0.46	0.78		0.15	0.11	0.37	0.13	0.25	0.24	0.14	0.10	0.34	0.09	0.11	0.11	0.09	0.09		0.13	0.16	0.13	0.21	0.50	
(e)Wrong dialling pulse Rec.by MFCR	0.66			3.26	0	0.48	0.24	0.34	0.05	0.03	0.08	0.11	0.09	0.01	0.02	0.04	0.31		0.16	0.28	0.2	0.24	0.03	
(f)Vacant code dial	7.24	8.74		10.27	5.99	7.67	7.66	9.66	9.23	9.29	9.30	9.38	8.91	8.97	8.85	9.44	9.93		8.86	9.68	9.06	9.03	8.29	
(g)Subs.Abandons in the Mid of dial.	36.05	36.45		28.93	41.25	38.74	42.07	33.38	37.96	36.83	36.73	35.19	35.54	34.37	35.06	35.59	35.96		37.24	34.93	35.37	34.27	33.85	
(h)Interdigit pause	5.42	9.07		12.62	24.38	23.21	12.24	11.29	3.89	3.86	2.81	2.51	2.36	2.44	2.8	3.23	3.14		3.39	3.61	3.30	3.04	3.21	
(i) b+ c+ d+ e	1.17	0.78		3.52	0.11	1.01	0.77	0.73	0.31	0.36	0.19	0.46	0.19	0.13	0.13	0.13	0.48		0.32	0.47	0.42	0.52	0.55	
4.Sender Time out	1.53	1.12		1.49	1.11	0.50	1.18	1.24	2.01	2.95	3.12	2.88	2.35	2.56	2.26	1.97	2.1		4.41	2.16	2.53	2.48	1.95	
5.DPIR Trunk hunt N.G.	0	0.0		0.33	0.0	0.0	0.0	0.0	0.0	0.03	1.54	4.30	5.42	5.10	2.81	0.28	0.49		0.41	0.02	0.01	0.0	0.0	
6.No. of call matured.	35.99	37.28		27.21	18.60	20.43	24.67	38.86	35.87	32.72	28.11	25.22	23.09	24.02	26.49	29.66	29.86		26.80	30.45	29.49	28.76	36.12	
7.Others.	2.84	2.17		0.0	3.95	5.27	8.42	4.32	2.27	2.69	5.16	5.72	5.33	5.47	4.36	3.00	3.26		4.86	3.84	6.03	7.89	0.0	

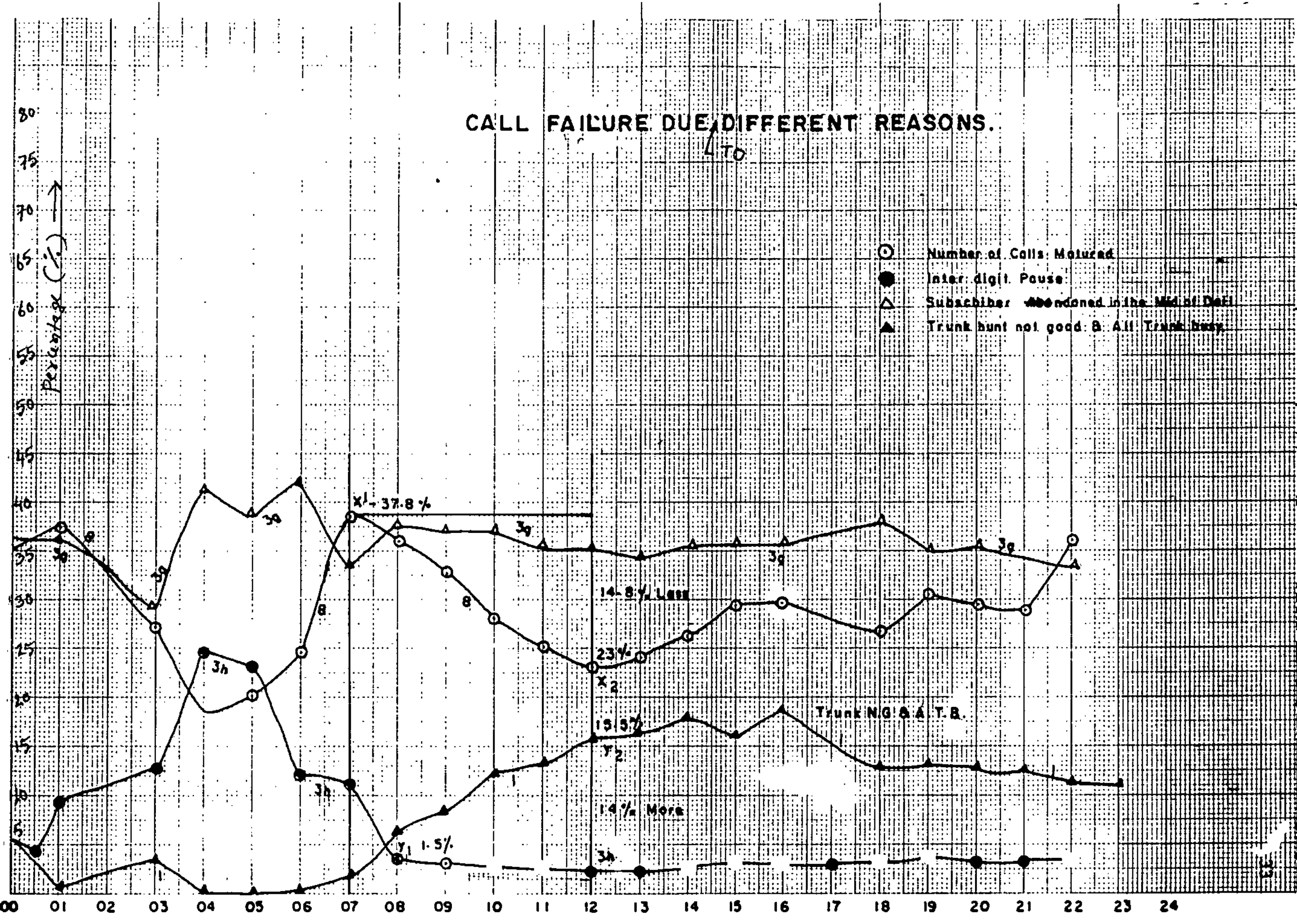


Fig. 3.3 : Call failure due to different reasons.

Table: 3.2 - **Hourly Variation of Offered and Succssuful Calls in Dhaka TAX.**

	0.00hr.	1.00 hr.	2.00hr.	3.00hr.	4.00hr.	5.00hr.	6.00hr.	7.00hr.	8.00hr.	9.00hr.	10.00hr.	11.00	12.00	13.00	14.00	15.00	16.00	17.00	18.00	19.00	20.00
	Calls offered 1 / Successful calls 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2	1 / 2
4.9.87									1873 / 391	3119 / 615	4209 / 782	5016 / 845	4975 / 913	5077 / 912	4833 / 985	4621 / 969	4420 / 880	4952 / 975	4515 / 867	4712 / 868	4910 / 889
5.9.87	3747 / 833	1650 / 388	813 / 156	571 / 97	380 / 79	236 / 73	339 / 96	766 / 193	1917 / 329	2747 / 558	4205 / 721	5177 / 779	5557 / 743	5663 / 880	6096 / 839	4898 / 815	4600 / 872	4938 / 822	4327 / 793	4383 / 904	4953 / 850
6.9.87	3104 / 715	1557 / 380	666 / 165	529 / 109	202 / 48	296 / 57	392 / 105	768 / 199	1907 / 316		5365 / 729	5965 / 844	7282 / 919	8811 / 915	8867 / 910	7197 / 894	6903 / 970	5666 / 938	5687 / 954	5585 / 902	6559 / 897
7.9.87	3976 / 804	2090 / 412	1077 / 233	655 / 110	403 / 98	292 / 64	294 / 94		1766 / 395	3827 / 710	5530 / 843	6523 / 905	7354 / 912	9032 / 985	1055 / 1126	8560 / 1118	7632 / 1188	7227 / 1114	6748 / 939	6167 / 936	6721 / 924
23.7.87	3594 / 783	1667 / 354	809 / 177	421 / 91	439 / 73	244 / 49	404 / 124	905 / 212	1246 / 350	2956 / 628	4979 / 738	6334 / 760	5168 / 669	6412 / 833	4612 / 769	4471 / 813	4465 / 809	3954 / 689	3336 / 621	3639 / 643	404 / 815
24.7.87	2593 / 660	1637 / 363	758 / 198	511 / 104	493 / 109	272 / 73	451 / 92	714 / 169	1516 / 346	2605 / 593	3557 / 820	4765 / 199	5208 / 862	4756 / 856	4739 / 970	4610 / 880	4254 / 796	4847 / 945	4257 / 836	4463 / 988	4676 / 973
25.7.87	3338 / 810	1775 / 440	986 / 140	617 / 139	288 / 81	256 / 58	431 / 95	819 / 193	1904 / 552	3394 / .667	5498 / 859	6528 / 992	6632 / 1001	7135 / 1087	6944 / 980	7239 / 980	7328 / 1180	6356 / 1012	6100 / 949	5628 / 863	5570 / 845
26.7.87	3870 / 700	1891 / 480	839 / 221	384 / 114	357 / 76	258 / 52	586 / 92	919 / 164	1661 / 361	3086 / 600	4721 / 729	5920 / 846	6622 / 828	6491 / 893	6501 / 890	7193 / 899	6796 / 921	6758 / 906	5883 / 835	5336 / 788	6057 / 906
27.7.87	4968 / 690	2248 / 446	982 / 196	765 / 116	379 / 73	285 / 70			1661 / 363	4073 / 754	5105 / 874	6365 / 943	7336 / 948	7336 / 1055	7454 / 1067	8557 / 1018	7752 / 1081	7462 / 1095	6323 / 1100	5737 / 1015	5469 / 900

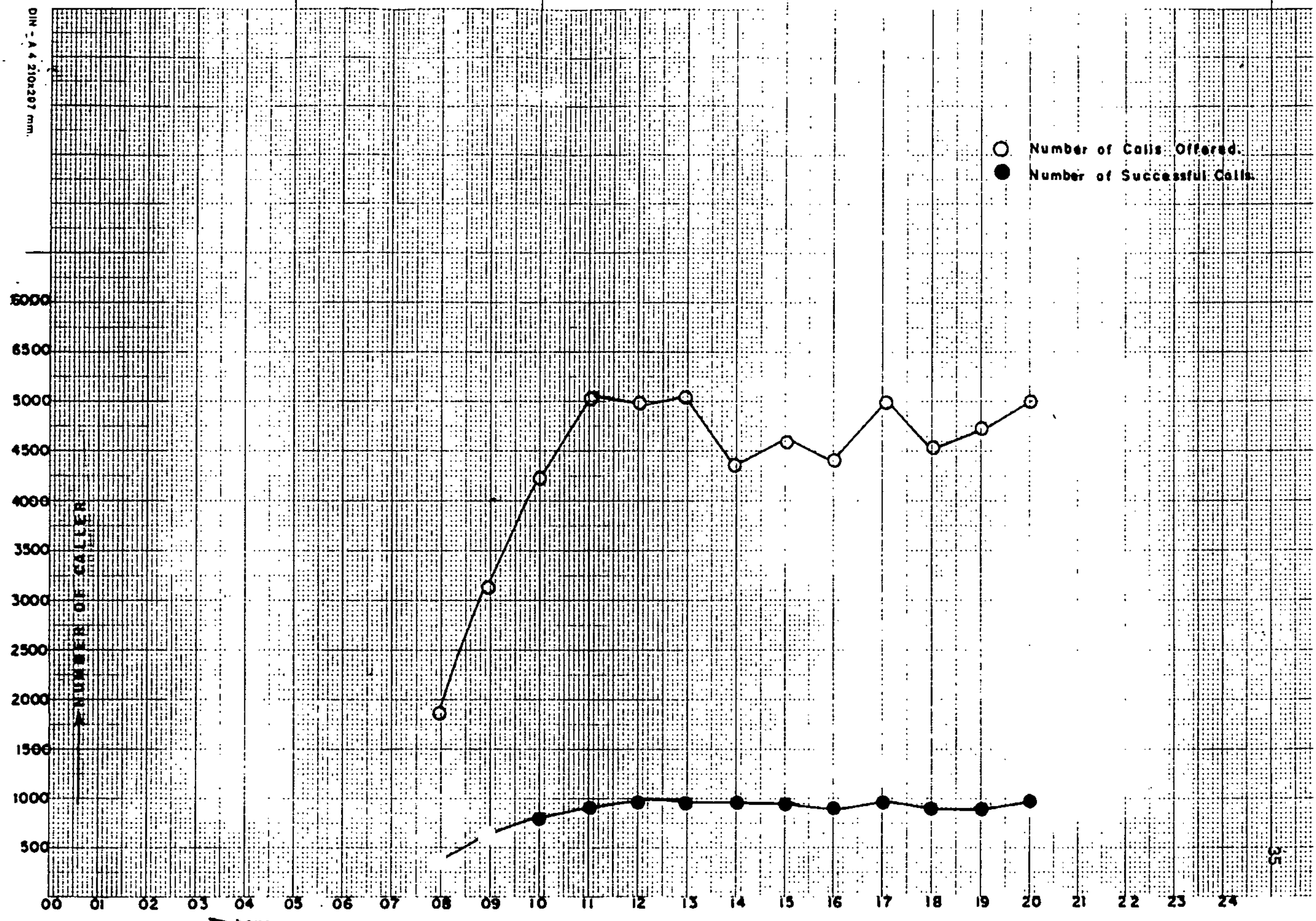


Fig. 3-4 Hourly Variation of offered and Successful calls in Dhaka TAX. 4.9.87.

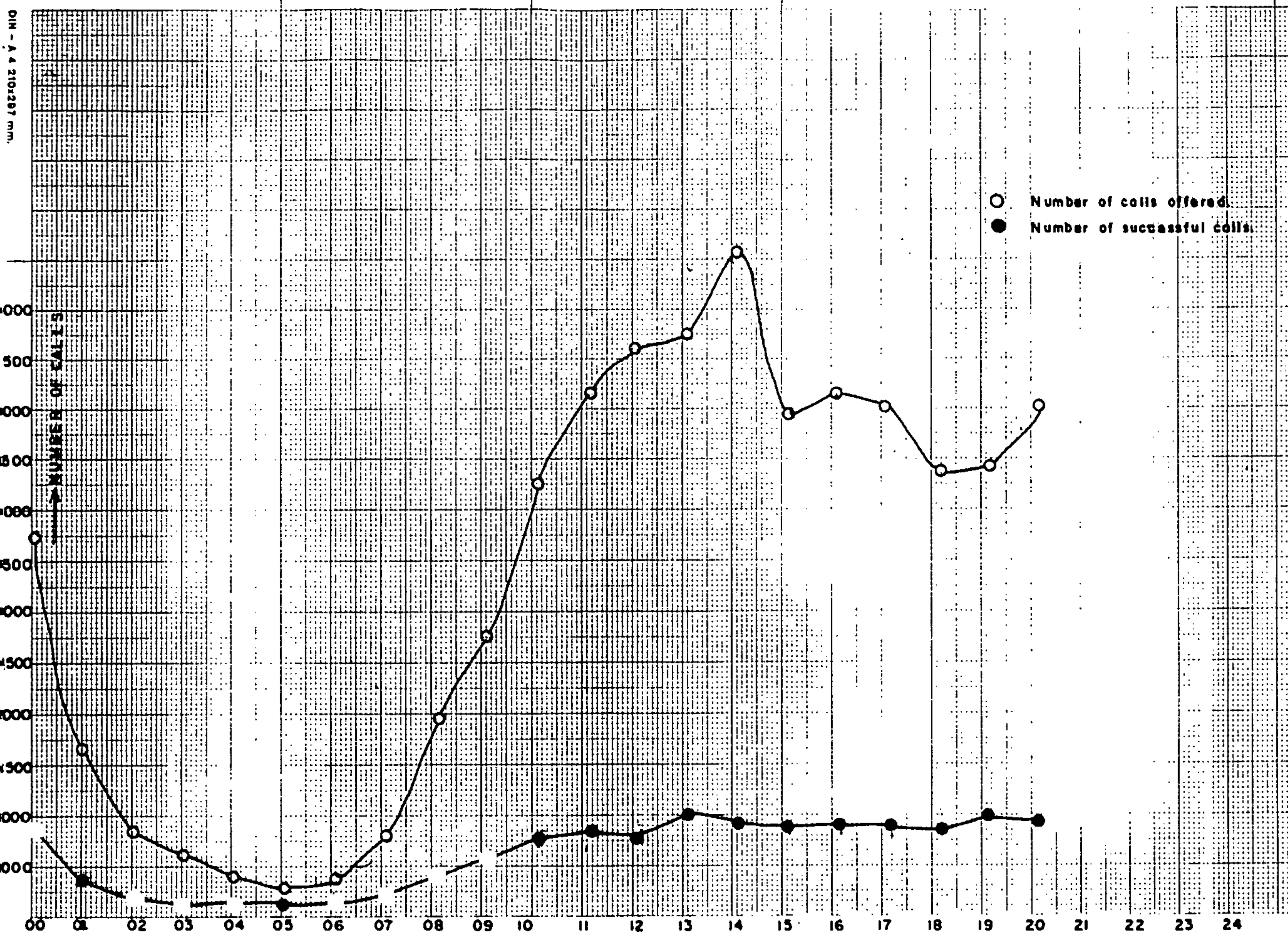


Fig:3.5– Hourly variation of offered and successful calls in Dhaka TAX on 5·9·87

DIN – A 4 210x297 mm.

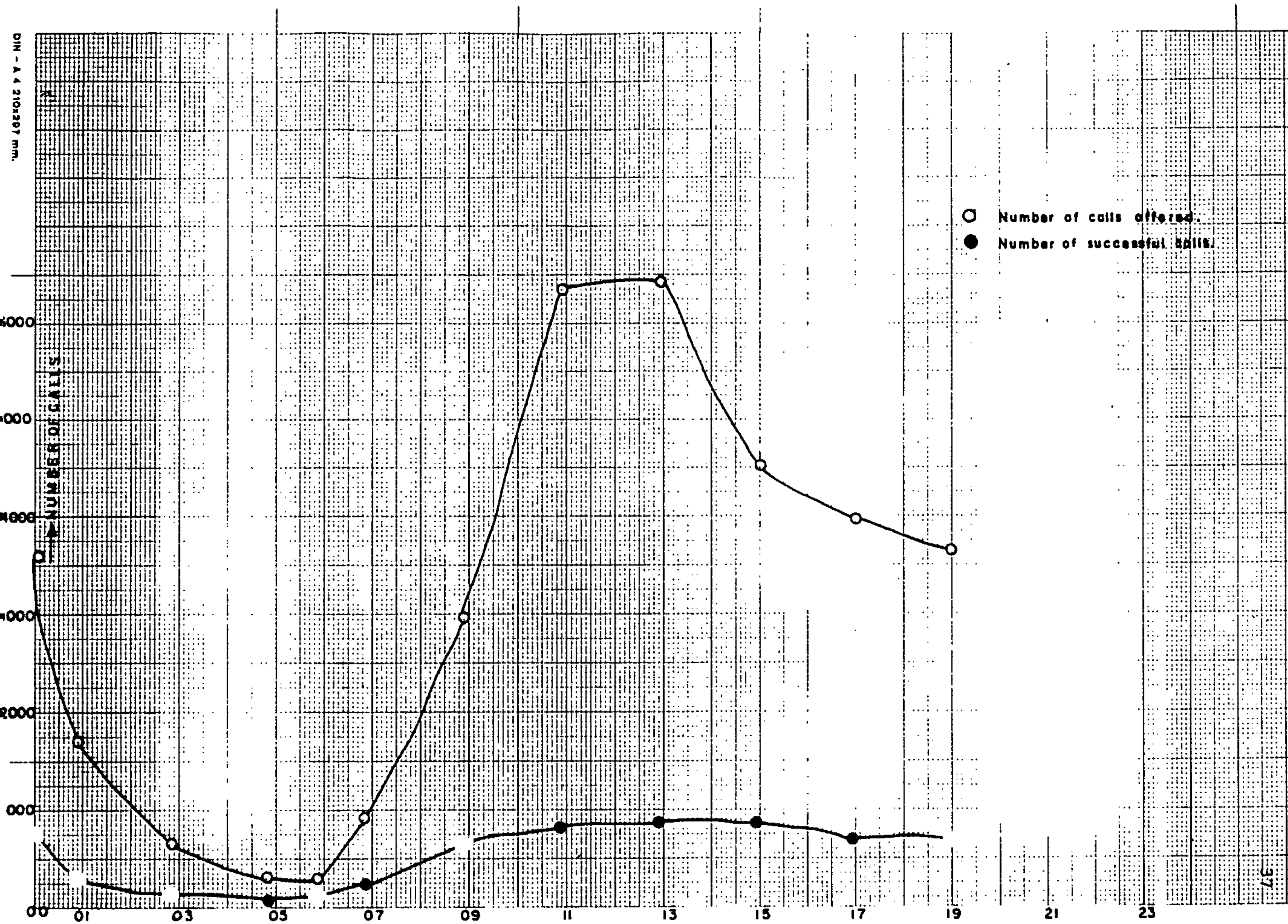


Fig.3.6 : Hourly variation of offered and successful calls in Dhaka TAX on 23·7·87

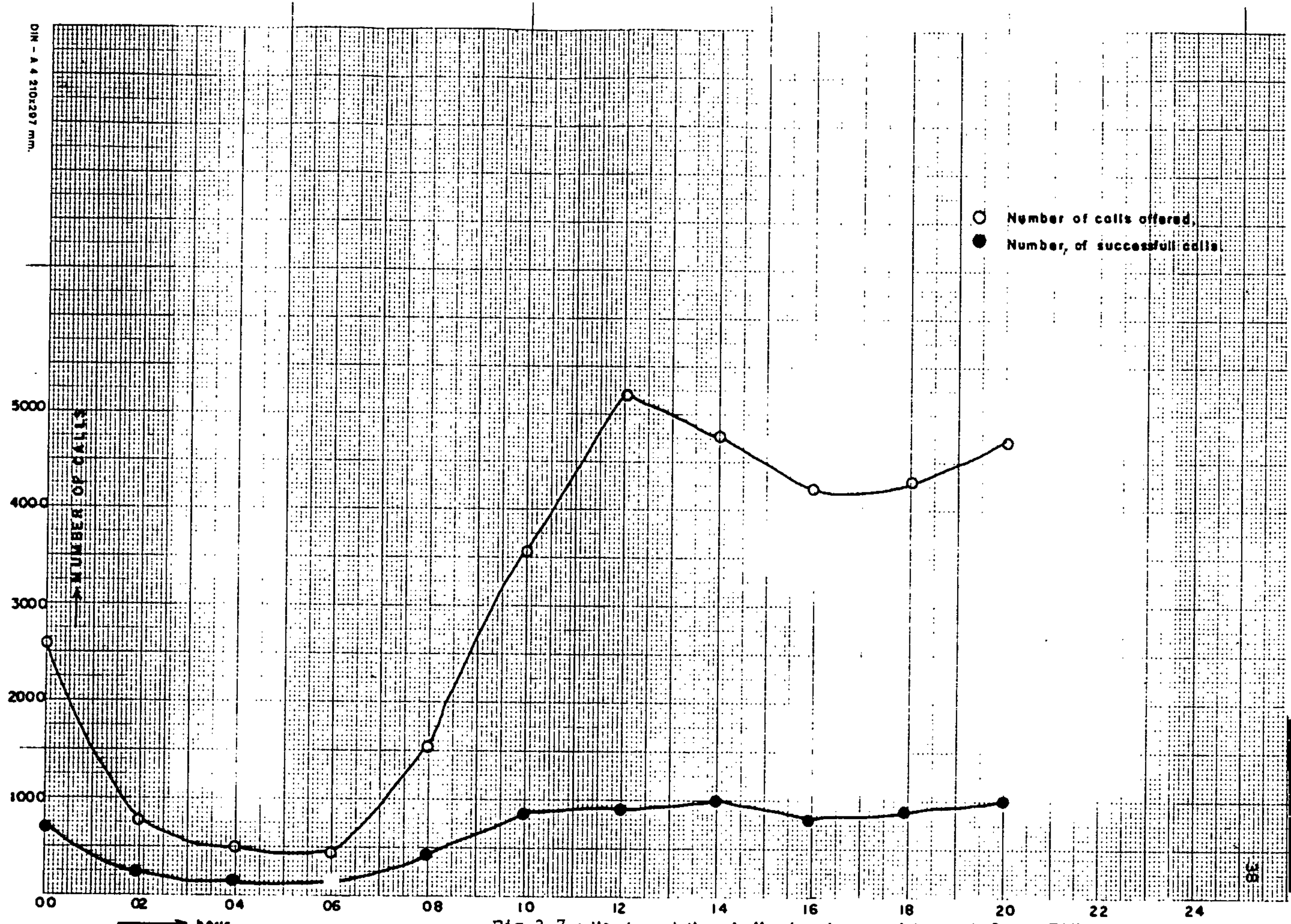


Fig.3.7 : Hourly variation of offered and successful calls in Dhaka TAX. 24.7.87

3.3 Problems with multiplexing of analogue signals:

The following problems are noticed in multiplexing and demultiplexing process of analogue signals:

i. Inherent thermal and shot noise.

ii. Crosstalk due to improper filtering.

iii. Intermodulation noise due to non-Linearities.

iv. High cost involvement.

v. Less channel capacity compared to PCM system.

3.3.1 Thermal noise and shot noise

Thermal noise is the noise occuring in all transmission media and in all communication equipment arising from random electric motion. It is characterised by a uniform distribution of energy over the frequency spectrum and a normal (Gaussian) distribution of levels.

Every equipment, element and the transmission medium contribute thermal noise to a communication system, provided the temperature of that element of medium is above absolute zero. Thermal noise is the factor that sets the lower limit for the sensitivity of a receiving system. Often this noise is expressed as a temperature referred to absolute zero (kelvin).

Thermal noise is a general expression referring to noise based on thermal agitations. Thermal noise is directly proportional to bandwidth and temperature. The amount of thermal noise to be found in 1 Hz of bandwidth in an actual device is

$$P_n = KT \text{ (W/Hz)}.$$

Where K = Boltzmann's constant. = $1.3803X(10^{-23})$ J/°k

T = absolute temperature (°K) of thermal noise.

For a band limited system (i.e. system with a specific bandwidth).

P_n = KTB (W)

where B= Bandwidth (Hz).

Shot noise is due to the discrete nature of electron or other charge carriers flow and is found in most active devices. This type of noise involves random fluctuations about an average particle flow. For example, electrons flowing between cathode and anode in a vacuum tube, electron and holes flowing in a semiconductor, Photoelectrons emitted in photodiodes etc. Although averaging over many particles the flow is found constant, there will be fluctuations about this average. This fluctuation causes a noise voltage to develop. The mechanism of fluctuations depends on the particular process. From statistical analysis it is observed [22] that mean squared fluctuation about the average value is proportional to the average value it-self.

In multiplexing thermal noise is the inherent noise present in all resistors and conductors used in the circuitry. Besides, shot noise is present in the active devices like transistors, diodes etc. used in the multiplexing and demultiplexing equipment.

3.3.2 Crosstalk:

Crosstalk is defined as the disturbance created in one communication circuit by the signals in other communication circuits. It is the disturbance created by another circuit due to circuit non-linearities, transmittance and electromagnetic coupling. In otherwords crosstalk refers to unwanted coupling between signal paths. Essentially there are three causes of crosstalk:

a) The first is the electrical coupling between transmission media, for example, between wire pairs on a VF cable system.

b) The second is poor control of frequency response i.e. defective filters or poor filter design.

c) The third is the non-linearity performance in analog(FDM) multiplex system.

In multiplexing and demultiplexing process crosstalk arises due to improper filtering. This is shown in fig. 3.8 where message signals have a band of frequencies from 0 to 4 KHz, actual frequency band is from 300 to 3400 Hz. The extra band 3400 Hz to 4000 Hz is used for signalling purpose. Now when 0.3 to 3.4 KHz VF signal is modulated with 16 KHz carrier as shown in fig. 3.8, the AM wave produced contains 16.3 to 19.4 KHz (upper side band), 12.6 to 15.7 KHz (lower side band), the original VF signal and other products of modulation. To transmit single side band (upper side band in this case), band pass filter (having transmitting band of frequencies from 16 to 20 KHz) is used. This filter supresses other products. But since, no ideal filter is available, signal frequencies will be slightly below 16 KHz and slightly above 20 KHz. These unwanted signals will fall in the channel '1' and channel '3' band causing crosstalk. Similarly, crosstalks are present in group and supergroup stages.

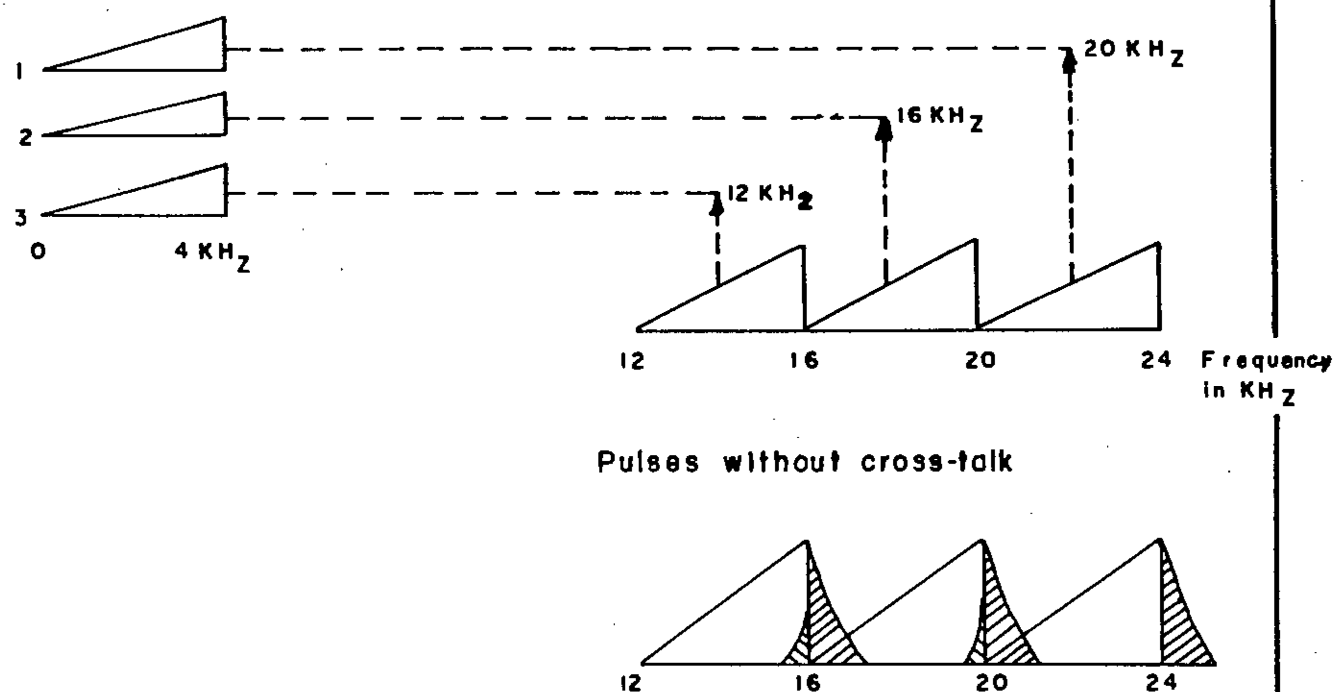


Fig -3.8 : Cross talk due to improper filtering

3.3.3 Intermodulation noise due to non-linearities:

Intermodulation noise is the result of the presence of intermodulation Products. In other words undesired modulation caused by the nonlinear characteristics of components is known as 'intermodulation noise. It produces new frequencies which include sum and difference products of the original frequencies, harmonics of the original frequencies and sum and difference products of the harmonics. There are various types of non-linearities that can cause intermodulation e.g.

i) amplitude non-linearity i.e. with change in input level, amplifier gain will change.

ii) frequency non-linearity i.e. for different frequency input of same amplitude, output amplitude will be different.

iii) Phase non-linearity, i.e. output phase will change according to the input amplitude.

iv) Phase and group delay, i.e. output phase will vary for different frequency input.

In multiplexing and demultiplexing equipment the main causes of intermodulation are:

i) incorrect operating point: This causes amplitude non-linearity and so intermodulation products.

ii) Overloading : Since linear range is limited, overloading drives the component in non-linear range i.e. amplitude nonlinearity occurs for overload condition.

iii) frequency distortion: This means variation of gain with frequency. The output waveform becomes distorted giving a noise. This effect can be minimised by using equalisers.

Besides, intermodulation distortion occurs in audio programme transmission. An audio programme signal is a complex wave consisting of many component frequencies. Normally its bandwidth is equal to 3 times the voice frequency bandwidth. If these frequencies are transmitted via equipment that produces intermodulation, new frequencies are produced. Some of these new frequencies will fall within the bandwidth of the transmitted audio programme, producing distortion which lowers the quality of the received programme.

3.3.4 <u>High cost involvement</u>:

In frequency division multiplexing (FDM) system thermally stabilised crystal controlled synchronized ocillators[22] with an accuracy of 1 part per million (PPM) are needed to generate the carrier signals. These devices are extremely expensive, but furtunately their cost may be shared between several multiplexers since the carrier frequencies remain identical. Still it is costlier than pulse code modulation (PCM) technique.

The filters used to remove the unwanted sidebands(multiplexer), and those used to filter the demodulated channel signals (demultiplexer) cannot be shared. They are difficult to design and are extremely expensive.

3.3.5 Less channel capacity compared to PCM

The bandwidth requirement for an FDM equipment designed to combine 12 telephone channels stretches from 60 to 108 KHz and uses transmission media which may be either a deloaded audio cable, open wire, or special pair cable. This multiplex provides the first order within the FDM hierarchy, and is the basic unit from which wideband signals representing larger number of chnnels are built up. The subsequent levels in the FDM hierarchy after the primary group are referred to as super, master and super-master groups. The primary group systems are used extensively on existing audio pair cables. But in PCM system 30 telephone channels can be provided on existing audio pair cables.

3.4 Existing cable characteristics:

Our objective is to use the existing cables for digital transmission. To attain this objective the characteristics of the existing cable should be known. The characteristics of the commonly used cables of BT&T Board are given here and measurement of different parameters of the cables of different routes were made.

Cables[23] in use are generally of PE-insulated, Laminated sheath type. The following diameters of cables are in use :

0.4mm (newly introduced):

Cross-Section A= .1257mm^2

Planning value : Loop resistance er Km = 290 ohms.

Planning value : dB Loss (800Hz) Per Km = 1.5 dB.

Mutual capacitance : 50 nF/km.

0.5mm (used earlier)

Cross section A = .1963mm^2

Planning value : Loop resistance per km = 190ohms.

Planning value : dB Loss (800Hz) per km = 1.25 dB

Mutual capacitance : 50 nF/Km.

0.6mm (used for local cables):

Cross-Section A = .2827mm^2

Planning value : Loop resistance per km = 130ohms.

Planning value : dB loss (800Hz.) Per km = 7.05 dB.

Mutual capacitance 52 nF/Km.

0.6mm (used for junction cables):

Cross-Section A = .2827mm^2

Planning value : Loop resistance per Km = 130ohms.

Planning value : dB Loss (800Hz) per Km = 0.95dB.

Mutual eapacitance : 42 nF/Km.

0.8mm (used for junction cables):

Cross-Section A = .5026 mm^2

Planning value : Loop resistance per Km = 75ohms.

Planning value : dB Loss (800 Hz) per Km = 0.70 dB.

Mutual capacitance 42 nF/Km.

0.9mm (used for junction cables):

Cross-Section A = .6361 mm^2

Planning value : Loop resistance per Km = 60 ohms.

Planning value : dB Loss (800Hz) per Km = 0.6 dB.

Mutual capacitance 34nF/Km.

The cable characteristics of some important routes are given in tables 3.3-3.8. Measuring techniques are discussed in appendix-A.

Table 3.3 Cable data of Shere-e-BanglaNagar - Mirpur route

Insulation : Lead sheathed Distance 8.4 Km. No. of joints : Not known

Date of measurement : 1986.9.12

Unit No.	Pair No.	Insulation		Capacitance nF/Km.	Loop Resistance Ohms/Km	Attenuation dB/Km.	Remarks
		A-Earth Mega-ohms	B-Earth Mega-ohms				
507/1	1	5	9	35.5	76.2		Cable Specification.
507/1	20	Break	-	-	-		a)150/0.9/LS.
507/1	21	30	10	35.7	73.8		b)The cable is not Loaded but balanced.
507/1	40	7	10	35.7	77.4	not measured	
507/2	1	80	80	25.8	75.0		c)Pressurized.
507/2	20	4	45	35.7	80.9		d)Direct burial.
507/2	40	16	11	36.3	72.6		e)Due to high dB Loss above 500 KHz attenuation could not be measured at 1 MHz.
507/3	40	25	18	36.9	73.8		
507/3	50	25	50	39.3	73.2		

Table 3.4 Cable data of Shere-e-Bangla Nagar-Tongi route

Insulation Lead Sheath Distance 20 Km. No. of joints 150 nos.

Date of measurement : 1986.9 . 16

Unit No.	Pair No.	Insulation A-Earth Mega-ohms.	Insulation B-Earth Mega-Ohms.	Capacitance nF/Km.	Loop resistance Ohms/Km	Attenuation dB/Km.	Remarks
497/4	1	70	70	17.5	60.5	Not	Cable specification:
497/4	50	15	20	15.8	62.5	measured	a)308/0.9/LS
497/5	1	9	8	14.5	62.5		b)The cable is not Loaded but balanced.
497/5	50	100	30	14.5	62.5		c)Pressurized.
497/6	10	80	25	13.0	62.0		d)The cable has passed through duct from Shere-e-Bangla Nagar Exchange to Amtali and direct burial from Amtali to Tongi.
497/6	50	150	55	16.5	59.0		
497/6	70	30	50	14.5	59.5		
							e)Attenuation could not be measured due to long distance.

Table 3.5 Cable data of Moghbazar – Gulshan route.

Insulation Polythelene (PE) Distance 6.2 Km. No. of joints : 25 Nos.

Date of measurement: 1986. 10. 03

Unit No.	Pair No.	Insulation A-Earth Mega-ohms.	Insulation B-Earth Mega-ohms.	Capacitance nF/Km.	Loops resistance Ohms/Km	Attenuation dB/Km.	Remarks
89/4	50	500	1000	31.0	126.45	12.0	Cable specification:
89/3	50	250	500	30.6	126.3	12.0	a)800/0.6/PE
89/2	50	200	175	30.48	126.45	11.8	b)The cable is not loaded but balanced.
89/1	50	400	1000	30.2	126.45	11.8	c)Not pressurized.
89/5	2	400	300	33.9	77.4	9.2	d)The cable is direct burial from Moghbazar to Amtali and ducted from Amtali to Gulshan.
89/5	52	250	200	33.9	77.6	9.2	
89/6	1	190	300	34.0	77.7	9.2	
89/6	51	150	200	33.9	78.1	9.2	
89/7	96	500	1000	33.9	77.9	9.2	
89/4	1	50	300	31.9	125.0	13.9	
89/10	50	100	23	31.1	124.8	13.7	

Table 3.6 Cable data of Central - Narayanganj route.

Insulation : Polythelene (PE) Distance 18.2 Km. No. of joints 75 Nos. (approx)

Date of measurement : 1986 . 10 . 18

Unit No.	Pair No.	Insulation		Capacitance nF/Km.	Loop resistance Ohms/Km	Attenuation dB/Km.	Remarks
		A-Earth Mega-ohms.	B-Earth Mega-ohms				
233/1	5	30	4	34.6			Cable specification:
233/1	45	100	4	34.3	not measu-	not	a)400/0.9/PE.
233/1	55	18	20	34.7	red	measured	b)The cable is not Loaded and balancing does not exist.
233/2	5	20	50	34.2			
233/2	25	5	4	33/9			c) Pressurized.
233/2	95	4	6.5	33.4			d) Direct burial.
233/3	25	30	9	34.1			
233/3	45	4	10	33.7			
233/3	95	40	4	33.4			
233/4	25	4	4	34.1			
233/4	45	50	4	33.5			

Table 3.7 Cable data of Moghbazar-Shere-e-Banglanagar route

Insulation : Polythelne (PE) Distance 3.6 Km. No. of joints 21 .

Date of measurement: 1985. 7 . 11

Unit No.	Pair No.	Insulation A-Earth Mega-ohms.	Insulation B-Earth Mega-ohms.	Capacitance in nF/Km.	Loop resistance Ohms/Km	Attenua-tion dB/Km	Remarks
8/1	26	4.00	300	41.4	145.8	14.5	Cable specification:
8/2	35	300	400	41.4	147.8	14.6	a) 600/0.6/PE
8/3	20	120	140	41.7	147.8	14.5	b) The cable is not loaded but balanced.
8/4	7	110	120	41.7	146.9	14.6	c) The cable is
8/5	37	30	500	41.7	147.5	14.75	Pressurized.
8/6	2	125	125	41.4	147.2	14.6	
8/7	25	700	1200	41.7	147.5	14.6	
8/8	44	500	1000	42.2	147.2	14.6	
9/1	42	1000	1000	41.4	147.5	14.6	
9/2	13	250	250	41.7	146.4	14.8	
9/3	49	1000	1100	41.7	146.7	14.7	

<u>Table 3.8 Cable data of Central-Moghbazar route</u>

Insulation - Polythe ene (PE) Distance 3.7 Km No. of joints 12 Nos.

Date of measurement: 1986. 7. 19

Unit No.	Pair No.	Insulation		Capacitance nF/Km.	Loop resistance Ohms/Km	Attenua- tion dB/Km.	Remarks
		A-Earth Mega ohms	B/Earth Mega ohms				
85/6	68	500	2000	34.9	60.00	9.7	Cable specification:
85/7	74	500	800	34.6	60.2	9.7	a)400/0.9/PE.
85/8	31	400	800	34.9	60.2	9.8	b)The cable is not loaded but balanced.
86/1	45	100	75	33.8	60.0	11.0	c)Pressurized.
86/1	55	40	70	33.8	59.5	9.4	d)underground burial.
86/1	69	55	70	34.0	59.7	9.5	
86/1	85	5000	200	33.8	59.7	9.4	
86/2	3	55	200	33.8	60.0	9.5	
86/2	10	300	300	34.1	60.2	9.5	
86/3	68	250	200	71.6	60.2	11.1	
86/4	54	400	1500	33.8	60.2	9.9	

3.5 Discussions:

In this chapter Traffic overflow, problems with multiplexing of analog signal and existing cable characteristics have been studied. From the analysis of the available data of Dhaka TAX it is found that amount of matured call depends apparently on getting of the Trunk. If Trunk/Line capacity can be increased matured call will certainly be raised.

Secondly, different problems regarding the multiplexing of the analog signals are studied and found that it is affected by inherent thermal and shot noise, crosstalk due to improper filtering, intermodulation noise due to non-linearities and high cost involvement.

Thirdly, an extensive study of the characteristics of the cables of different routes of Dhaka telecommunication region was performed. These data will be helpful in designing routes for PCM communication. It will be pointed out in the next chapter that the existing problems of less trunk/line and problems of multiplexing can be avoided if digital transmission is successfully introduced in our network.

CHAPTER-4

QUALITY OF DIGITAL TRANSMISSION OVER THE EXISTING ANALOGUE ENVIRONMENT.

4.1 Introduction:

This chapter deals with PCM principles, Transmission quality of the cables, planning of a PCM system over the existing analogue environment in Bangladesh, Planning requirements, results of the investigation and the results of the existing cable network in Dhaka. The analysis was made for 30 channel PCM system, since it has the widest global coverage.

4.2 PCM Principles:

Pulse code modulation (PCM) is the common method of converting analogue signals into digital format (and vice versa) so that they may be conveyed over a digital line system or digitally processed (e.g. digital switching)[13,22,26]

The conversion is based on three major principles:

a) Sampling

b) Quantizing

c) Coding.

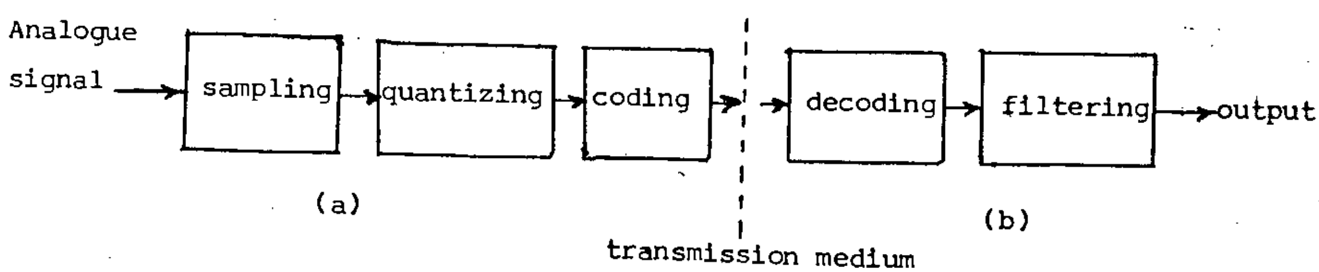


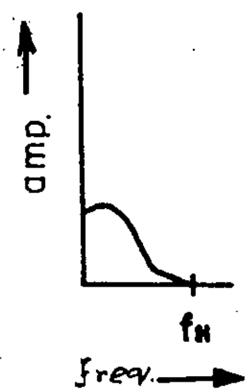
Fig.4.1 One channel PCM system; (a) transmitter, (b) receiver.

The principle of PCM Communication system is schematically shown in fig. 4.1

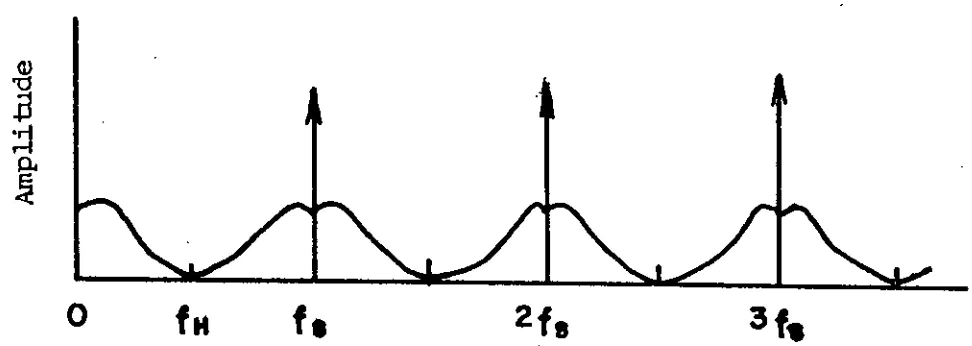
4.2.1 Sampling:

The transformation of a continuous signal waveform into time slots is called "sampling". The sampling theorem developed by Nyquist states that any waveform may be sampled and then completely reconstituted provided that the rate of sampling is equal to or greater than **twice the highest** frequency component of the waveform. Thus for a waveform limited to frequencies below f_H, the sampling rate f_s is given by $f_s \geqslant 2f_H$. The minimum value of $f_s = 2f_H$ is known as the Nyquist Rate of fequency for f_H. Generally, sampling below the Nyquist Rate will cause multilation of the reconstructed waveform due to the phenomenon of 'aliasing'.

Aliasing is the term given to the overlapping of the consecutive sidebands produced by the sampling process, which is essentially one of the amplitude modulation. This concept is illustrated in Fig. 4.2. It can be shown that sampling a wave-form at a rate of f_s produces upper and lower sideband versions of the waveform centred at frequencies of nfs (n = 1, 2, 3, ...). If fs $<$ 2 f_H , where f_H is the cut off frequency of the original waveform, the aliasing is produced, as illustrated. The diagram demonstrates that the minimum seperation of nfs and (n + 1) fs is $2f_H$ if aliasing is to be avoided. Clearly, sampling at $f > 2f_H$ ensure no aliasing but too high a rate is wasteful of bandwidth.



The waveform



The sampled waveform with $f_s = 2f_H$

Fig.4.2 ....

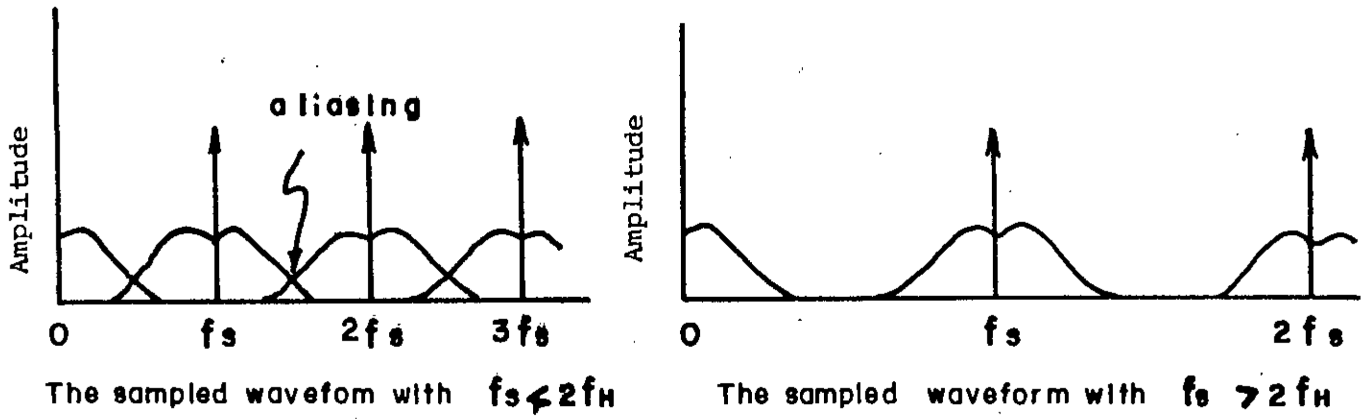


Fig. 4.2 Frequency spectra of sampled signal at,below and above Nyquist's rate.

Fig. 4.2 also illustrates that the original waveform may be retrieved by passing the sampled waveform through a low pass filter with a cut-off f_H. Fig. 4.2 shows in frequency domain the samples derived for a particular input waveform and the reconstituion of the samples. The telephony audio signal has a frequency power spectrum which is significant up to about 10 KHz. However, most of the power resides in the lower range. For economy of bandwidth telephony channels are band limited to 300-3400 Hz. In practice frequencies higher than 3400 Hz passed because of the skirt of the filter. To allow for this non-perfect filtering and hence keep aliasing negligilble CCITT recommended a sampling rate of 8KHz (i.e. once in every 125 μ sec), which is universally adopted to all PCM systems.

4.2.2 Quantizing:

Quantization is the process of measuring the value of each sample. The value is measured at the half way between two decision points. This point is known as quantization level. To examine the distortion arising from the fundamental requiement of P.C.M. that samples transmitted cannot be continuously variable but must be chosen from a finite set, the number of which is a function of the length of the binary number to assign for the transmission of each sample.

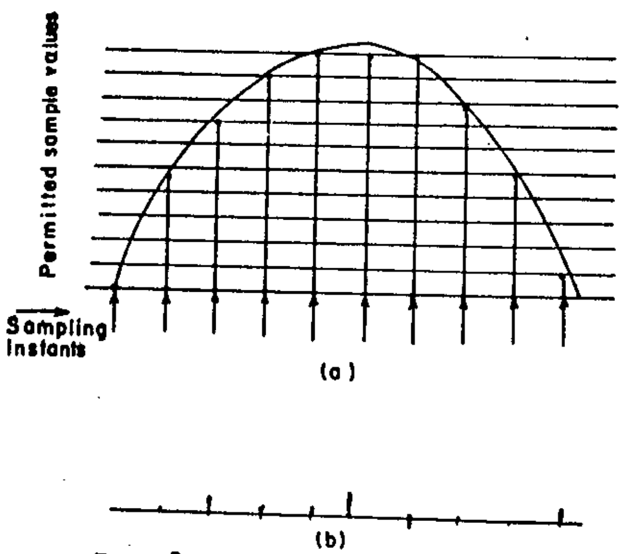


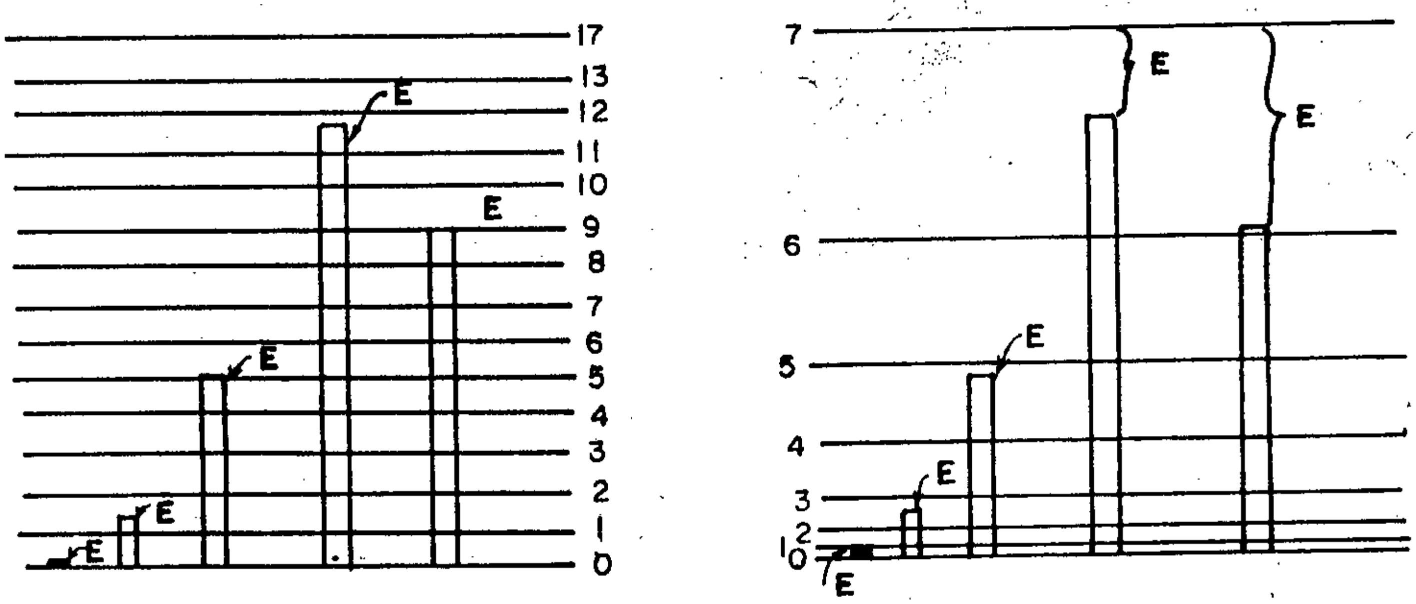
Fig. 4.3 Sampling of a section of wave form.
a. Sampled and quantised wave (assuming linear quantising)
b. Corresponding quantising errors.

Fig. 4.3 shows a section of wave-form being sampled. The permitted sampling approximations are indicated by the horizontal lines. It is assumed that the value selected is the nearest one + or — (in practice for reason of convenience the next lowest one is taken and at the recovery stage half a step value is added, which has identically the same result). The quantizing errors are shown below. It will be seen that the mean value is $\pm \frac{1}{4}$ S where S is the magnitude of one step. Except where there is a fortuitous correlation between sampling rate and transmitted frequency, and this occurs infrequently in speech, the mean power is represented by :

$$\int_{-\frac{1}{2}S}^{+\frac{1}{2}S} (x^2 dx) / S = \frac{1}{12} S^2$$

The random nature of this distortion results in a frequency content which is substantially flat over the pass band of the channel.

In another way quantising can be explained. After first amplitude limiting the audio signal each PAM sample is compared with a set of quantum levels and assigned to the next highest level. At the far end the PAM samples are reconstituted by generating sample heights equal to the appropriate quantum level. Clearly the rounding up process generally introduces an error E known as quantizing error as shown in fig. 4.4. If the sample steps are equally spaced then the quantising error to signal ratio is worse for all signals than for large signals. This problem is minimised by the use of spacing the sample thresholds Logarithmically so that Large signals have large error and small signals have small error thus giving essentially constant quantising error to signal ratio over the permissible amplitude range. This process also enables a wide range of amplitude to be encoded by a given number of quantising levels. Since the effect is the compression of this higher amplitudes into fewer quantising levels then the technique is also known as "companding".



Linear sample steps. Logarithmic sample steps.

Fig. 4.4: Companding of linear sample steps.

In a practical system either 128 or 256 quantum levels are used to represent both positive and negative samples using either the A-law or the μ-law Logarithmic approximations(as described in appendix-B).

4.2.3. Coding:

Coding is the expression of a quantised sample magnitude in terms of a binary number for transmission over the channel. Each of the quantum levels E is assigned a binary number. The first digit represents the sign of the sample : 0 for negative and 1 for positive. The remaining 6 or 7 digits then represent the magnitude—the first indicating whether upper or lower half; the second whether upper or lower quarter; the third whether upper or lower either , and so on. The single PAM sample representing one channel has now been covered to a PCM word of 7 bits (2^7=128 levels) or 8 bits (2^8= 256 levels), which may be transmitted at an arbitrary bit rate.

The reverse process, performed at the termination to the PCM system, is known as decoding.

In PCM 256 differently spaced quantizing intervals are provided in a practical system. 128 for the positive half and 128 for the negative half of the signal. $256 = 2^8$. So 8 bit binary code word is needed for the transmission of each sample. The first bit is polarity bit. 1 for +ve and 0 for -ve and remaining bits are absolute value bits. Fig. 4.5 below shows a 3 bit coding system.

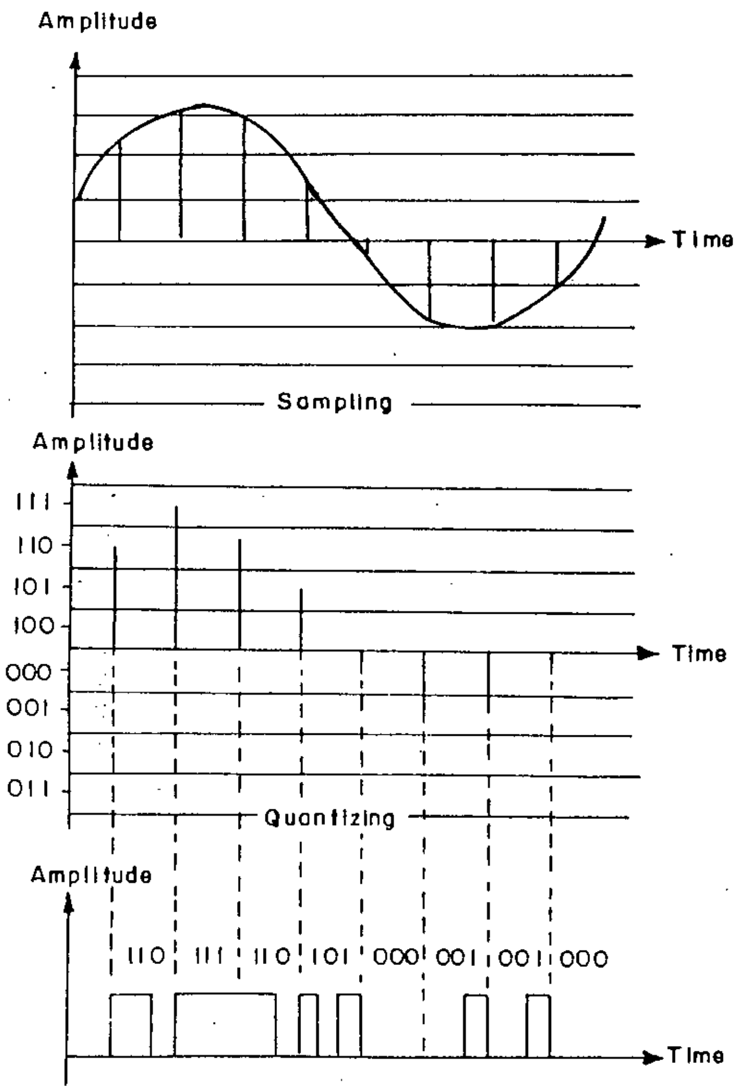


Fig. 4.5 : 3-bit Coding of a signal.

4.3 Transmission quality:

The quality of communication offered to the telephone user may be expressed in terms of frequency response, harmonic distortion, signal to noise ratio, and so on[22]. However, establishing values for these parameters is difficult since the term quality is imprecise and subjective. It is clear that the listener must be able to recognise the meaning of the spoken words and indeed the identity of the speaker. Furthermore, Prolonged conversations should be possible with only minimal listening fatigue. This is the minimum requirement.

Tests on human voice have been made at various Laboratories over the years, for example [7]. It is interesting to note that slightly different frequency spectra are obtained when the speech pattern of the European and Bangladesh male and female are analysed. Figure 4.6 illustrates the typical frequency spectrum of human voice.

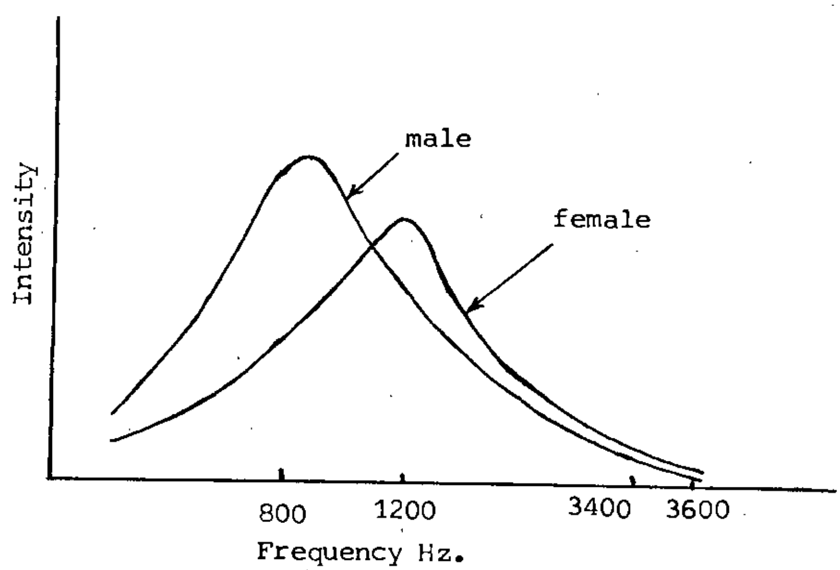


Fig. 4.6 : Typical frequency spectrum of human speech.

The internationally accepted telephone specification has the following characteristics: a frequency response of 300 to 3400 Hz, harmonic distortion better than 26 decibels and a signal to noise ratio better than 26 decibels and a signal to noise ratio better than 30 db.

4.3.1 <u>Frequency response or bandwidth:</u>

The frequency response of the ear (i.e. how it reacts to different frequencies) is a non-linear function between 30 Hz to 30 KHz; however, the major intelligence and energy content exists in a much narrower bands[28]. Tests have shown that low frequencies upto 600-700 Hz add very little to the intelligibility of a signal to the human ear, but in this very band much of the voice energy is transferred. For economical transfer of speech intelligence, a band much narrower than 20 Hz to 20 KHz is necessary. In fact the standard bandwidth of a voice channel is 300-3400 Hz as per CCITT Recommendations G.132 and G. 151A.

4.3.2 <u>Phase and attenuation constant:</u>

The propagation constant of a transmission line is given by

$$=\sqrt{(R + jwL)\,(G + jWc)} = \alpha + j\beta$$

where α is the attenuation constant and β is the phase shift constant. Both α and β are functions of frequency and geometrical shape of the line. ∴ The attenuation constant α follows a $\sqrt{f}$ curve at low frequencies. For a given cable inductance L, the value of WL will be also be greater than R and similarly WC will also be greater than G. This condition arises at frequencies above 60 KHz. At high frequencies, the attenuation curve tends to become linear but returns to the $\sqrt{f}$ characteristic as the frequency is further increased. This is due to skin effect at higher frequencies. Fig. 4.7 shows a typical attenuation curve.

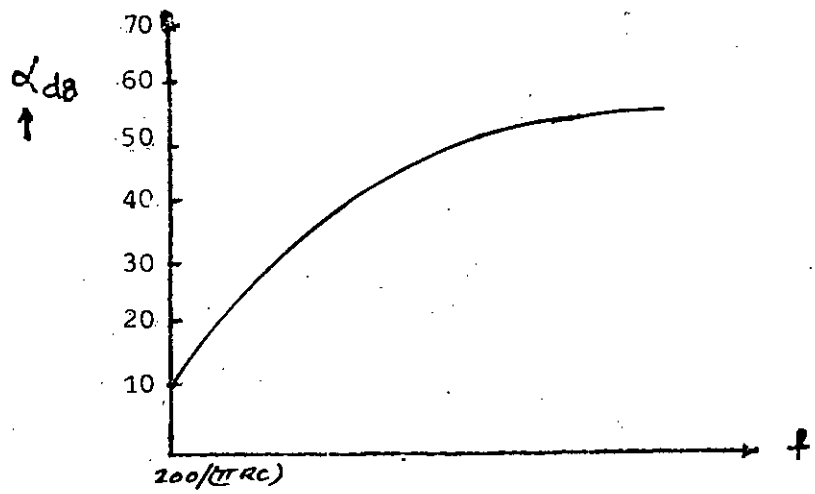


Fig. 4.7 : Attenuation characteristics of VF Cable.

The phase constant β also follows a $\sqrt{f}$ characteristics just as α. The attenuation constant α does not give a tone picture of the low frequency losses of the line. This is because :

a) The equipments at the two ends of the line are chosen to have an impedance equal to Zo, the high frequency characteristic impedance value of the cable. But,

b) The characteristic impedance of the cable is a function of frequency and is not equal to Zo at low frequencies.

Therefore, at low frequencies there is considerable impedance mismatch. That is, the losses at low frequencies include, in addition to the attenuation, the mismatch losses also;

a) The losses due to attenuation are called non-reflection attenuation.

b) The losses due to mismatch are called reflection attenuation and have two parts viz

i) attenuation due to mismatch at transmitting end.

ii) attenuation due to mismatch of receiving end.

Typical values of attenuation are

19 dB/Km at 1024 KHz for 0.6 mm cable.

15 dB/Km at 1024 Khz for 0.9 mm cable.

4.3.3 Crosstalk:

The definition of cross-talk is given in Chapter-3. In symmetrical lines (e.g. star quads) cross talk occurs because each line conducting voltage and current, produces an electro-magnetic field which transfer or couples a portion of energy carried by one line to another.

There are two types of cross-talk

i) Near end crosstalk (Next) :

This is a function of the coupling between pairs, measured at the same end of the cable as the signal source.

ii) Far end crosstalk (Fext):

This is a function of the coupling between pairs, measured at the remote end of the cable from the signal source.

It is to be remembered that where the disturbing line and the disturbed line carry signals in opposite directions, then 'Next' is the more serious form of crosstalk. The actual value for 'Next' can vary considerably between cables, although 68 dB may be regarded as a typical.

Mean value of α ranges from 50 to 85 dB.

When the lines carry signals in the some direction, the 'Fext' becomes the more serious source of crosstalk. A typical mean value for 'Fext' may be taken as 54 dB, for a range of value of 38 to 70 dB.

For effective transmission, there should be minimum coupling from one line to another. The degree of this coupling is often referred to as cross talk attenuation. It is defined as a ratio

$$a_n = 10 \text{ Log } \frac{P_1}{P_2} \text{ dB.}$$

a_n is the crosstalk attenuation

P_1 is power in the main line and P_2 is the power coupled to the next line.

The primary cause for cross talk is coupling between the lines in a cable. There are basically 3 types of coupling

a) Capacitive coupling.

b) Magnetic coupling.

c) Galvanic coupling.

4.3.4 Interference

Voice frequency (VF) cables are also affected by

i) Exchange interference.

ii) External interference.

Exchange interference:

This occurs in the vicinity of a switching equipment. The switching process may cause certain impulse noise spikes which can interfere with the signals carried by the VF pairs. The exchange interference depends on the cross-talk attenuation between the VF lines, number of active lines and on the type of exchange. Obviously, strowger type exchanges would cause more interference than crossbar and electronic exchanges.

External interference:

Lightning and power cables running parrallel to PCM routes are two major sources of external interferences. When the local terrain has a high resistivity and the route has a high lightening incidence, then the cables must be equipped with protective arrangement.

4.4 Planning of a PCM system over the Existing Analogue Environment in Bangladesh.

The symmetrical paired cables already installed for voice frequency lines can be economically utilized for PCM systems. Presupposition for that is to investigate the network concerning its ability to transmit signals of 2 M bit/s because originally cables were planned for voice frequency transmission. This necessary investigation can be done by analog as well as by digital measuring. The following important parameters for the use of PCM are listed below:

i) Overall loss from MDF to MDF between two exchanges in reference to 1MHz.

ii) Near end crosstalk.

iii) Number of simultaneously acting PCM-Systems.

Based on these data, one is able to start with the planning of how to bring PCM into action.

To put digital transmission systems into analog environment the following additional inquiries are essential.

i) Transmission plan.

ii) Level plan.

iii) Interface conditions.

iv) Structure of the network.

v) Integration of digital switches within digital transmission systems.

vi) Problems of stability, echo, singing etc.

4.4.1 Planning Requirements:

For reliable planning of digital transmission the following should be studied in detail.

Route and system parameters:

a. The length of the transmission route between two line terminating units.

b. Characteristics of the balanced cable:

   i) Conductor diameter.

   ii) Kilometric cable attenuation at 1 MHz.

   iii) Maximum cable temperature.

   iv) Value of the near-end cross talk to signal ratio.

   v) Associated standard deviation.

   vi) Number of 2 M bits/sec signals to be transmitted in a cable.

c. Any predetermined repeater locations.

d. It is to be assumed whether the pairs for both transmission directions are accommodated in the same cable (single cable

Operation) or the signals of both transmission directions being carried in separate cables (two-cable operation)

e. Are/any strong sources of extraneous interference present which influence digital signal transmission, line data, transmission etc. ?

there

f. Network preparation:

The following distorting networks have to be removed from the selected pairs

i) Loading coils.

ii) Longitudinal balancing networks (e.g. Longitudinal balancing capacitors).

iii) Capacitive build-outs.

iv) Other frequency dependent attenuating networks.

g. The planning should be made assuming that the individual regenerator sections do not exceed a bit error rate of 1×10^{-7} with a probability of at least 99%.

h. Balancing capacitors:

If it cannot be guaranted that all Longitudianl balancing capacitors have been removed, the maximum distance between the repeaters must be shortened by the corrective value ΔL approximately given by[27],

$$\Delta L = 0.4 \times d.$$

where d = conductor diameter.

ΔL = in Km.

i. Maximum starting section:

In the final stage of line planning, interference due to signalling pulses running in the parallel Voice frequency pairs of the same cable must be taken into account. These tend to increase the bit error rate of the digital transmission in general only in the immediate vicinity of the exchange. This influence can practically always be eliminted by shortening the first repeater section.

The length is determined as follows[27]:

$$\max L_s = L_{RS} \times 0.5$$

where L_s = Length of staring section.

LRS = Length of repeater spacing.

The starting section is 50% of a normal repeater section every time, because in this section near the line termination equipment the main problems are cross-talk and impulse noise.

Length of Repeater spacing (LRS) :

The length of repeater[14] spacing is determined as

$$LRS = \frac{aR}{ao.}$$

aR = maximum section loss.

a_o = cable attenuation in db/Km.

j. Power feeding:

The power feed length L can be calculated on the basis of the known repeater spacing. The voltage drop in different cables are given below for I = 50mA (const).

Diameter of the cable(mm)	Voltage drop (v)
0.4	9.66
0.6	3.51
0.8	1.98
0.9	1.53
1.2	0.90
1.4	0.66

The feeding dropped by the repeaters are fixed at a maximum of 13.5 volts.

Power feeding for the repeaters can be done either from one end or from both ends.

The power feed voltage is 100V or 200V can be selected depending on the national regulations for personal protection.

k. Number of cable pairs:

Each PCM system requires two pairs, and a PCM transmission line requires some additional pairs for spare lines, supervisory lines and service lines. Pair utilisation for PCM can not reach 100% because of near end crosstalk restrictions[23].

l. The type of the cable : (Layer type or unit type).

The rules to assign the pairs used for PCM differ according to the cable type. The rules are based on the evaluation of the average crosstalk of each cable type, so that no crosstalk measurement are needed during installation of PCM equipment.

m. Selecting suitable pairs:

The transmission quality of the digital line path can be significantly affected by selecting suitable pairs for the two transmission directions. The aim of this selection is to achieve the

the greatest possible near end crosstalk attenuation betwen pairs of different transmission directions.

The scope for selecting pairs may be restricted however by the way in which the individual production lengths of a cable are spliced together.

The favourable method is systematic splicing, i.e. identical terminals are connected together. Random splicing reduces near-end crosstalk attenuation and means that the spacing between regenerators has to be reduced. When systematic splicing is employed, it is advisable to route the two transmission direction in pairs which are widely separated or in two separated cables.

n. Utilisation of existing facilities:

If a number of cable pairs are loaded, the line may be configurated to allow installation of loading coils, which can be utilized economically for the repeater installation. In this case, actual loading sections are chosen as the repeater sections.

The nominal loading section specified by an H-code(US specification), is 1.83 Km and this spacing may be reasonable for a repeater spacing of PCM line over a normal cable.

4.4.2 The Parameters required for the Investigation:

The following cable data will be required for calculating the various parameters of digital communications:

i) a_{CT} = Near end crosstalk value in db. This value can be measured as per Fig. 4.8.

Test circuit:

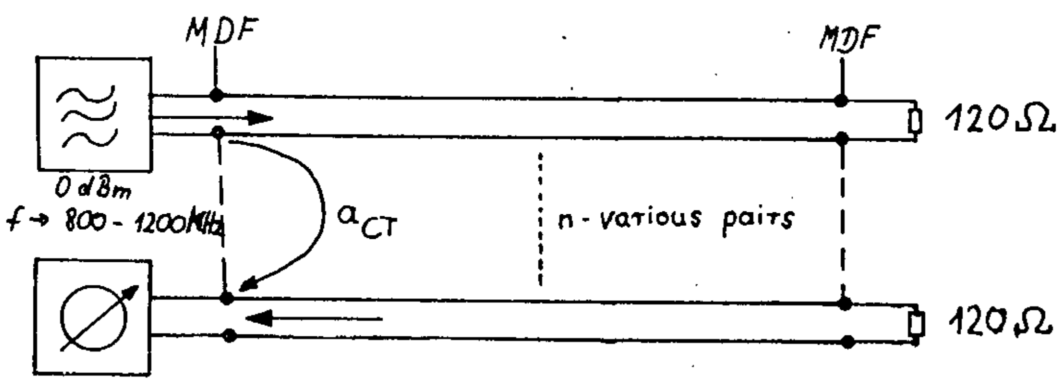


Fig. 4.8 : Measurement of Near end Crosstalk.

The near end cross talk attenuation is very important value. The minimum value of a_{CT} should therefore be found within the frequency range of 800 to 1200 KHz. The a_{CT} value of a 0.9mm, 400 pair cable was found to be 53.5 db. Practical values might be 60 to 70 dB.

ii) a_o = Cable attenuation in dB / Km.

The cable attenuation must be measured at 1024 KHz. The value of attenuation is given per Km.

Cable attenuation[27] a_o per Km at 1 MHz as a function of the mutual capacitance C and conductor diameter d is referred to a cable termperature of $T_o = 10^oC$. The following equations are applied depending on the cable dielectric :

For PE cable[13], $a_o = 0.22 \times \frac{C'}{d}$

The value is about 15 db/Km in case of a.6mm cable and 11 db/Km in case of 0.9mm PE cable.

iii) a_{sn} = Signal to noise ratio

A signal to noise ratio (a_{sn}) of 25 db is required (800 to 1.2 MHz) at the input of a regenerator of a bit error rate (BER) 1×10^{-7} is to be achieved for a line section with a maximum of 32 regenerators. An error rate of 10^{-6} has been recognised[23] as tolerable for high quality PCM telephone transmission, although this value has not yet been fixed internationally.

iv) q(s.N) = sum related allowance is a factor dependent on standard[27] deviation and number of PCM systems to be connected in parallel. For 5 systems the value is 10 db and for one system the value is 0 db. practical values lies between 10 to 15 db.

v) a_R = Maximum section loss :

The maximum repeater section loss (a_R) achievable in terms of nearend crosstalk is calculated from [13].

$$a_R = a_{CT} - a_{sn} - q\,(S.N).$$

Maximum repeater section loss should be confined within 40 db.

4.4.3 Calculation of the Number of Repeaters Required.

For implementation of PCM transmission in different sections of the existing analogue networks without changing the present cables, it may be necessary to adjust the number of repeaters on the basis of the average cable attenuation and maximum section loss allowable between two PCM repeaters. The average cable attenuation of the existing system is obtained by averaging over the measured data i.e.

$$a_o = \frac{1}{j} \sum_j a_{oj} \qquad (\text{dB/Km})$$

where j is the number of measurement.

The maximum allowable section loss a_R (dB/ km)between two PCM repeaters has been defined in the previous section. In terms of a_o and a_R, the length of repeater spacing (LRS) is given by

$$LRS = \frac{a_R}{a_o} \qquad (\text{Km})$$

For the first two repeaters at the starting sections between two exchanges the length of the starting section (LS) should be

$$LS = \tfrac{1}{2}\, LRS \qquad (\text{Km})$$

Let the distance between two consecutive exchanges be L (Km). Then the number of repeaters required is given by

$$R = \frac{L - LRS}{LRS} + 1$$

Calculations for different major routes are shown in appendix-C.

4.5 Economy of PCM Systems:

Frequency Division Multiplexing (FDM) techniques for telephony primarily suited for Long-haul transmission, are often uneconomical[23] for short distances. In some countries, small-channel capacity systems on symmetrical paired cables do not prove economical for distances of less than about 15 to 20 km, where the inter-ffice trunk line distribution is often highly concentrated. The reasons are that in FDM systems the costs for the terminal equipment dominate particularly for short-distance transmission.

Signalling equipment associated with the telephone channel makes up a significant part of terminal equipment costs. For FDM systems, this signalling equipment is usually separate and constitutes a substantial part of the total per-channel cost. In PCM transmission, signalling can be conveniently incorporated with the primary multiplex equipment.

The channel filter of an FDM system makes up about 20 to 35% of the total terminal cost. In PCM systems, however, an expensive channel filter is not required.

If the system cost comparison is considered, it is preferable to study the cost of the network including switching and subscriber's terminal equipment. According to "Documents and proceeding of the study day on Impact of Digital facilities on planning of Telecommunication Network[13]" presented by Sweden to CCITT's planning committee in 1980, the cost distribution per subscriber's line in the traditional analog network and the estimated cost distribution per subscriber's line when such analog network is employed to the digital network is estimated as in Figs. 4.9 and 4.10. From fig. 4.10 it can be observed that the cost for digital installation is 27% less compared with an analogue installation for identical channel capacity.

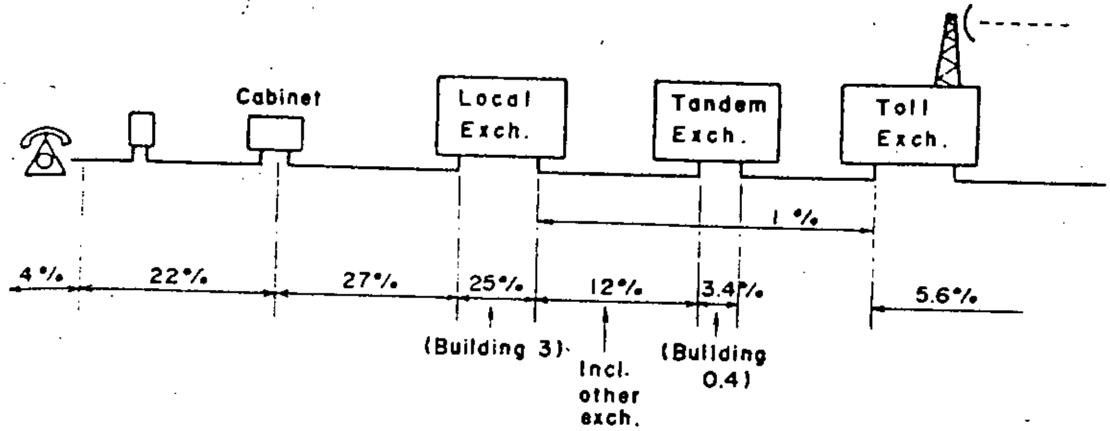


Fig.4.9 *Example of Cost Distribution for One Subscriber Line In Traditional Analog Network (Total = 100%)*

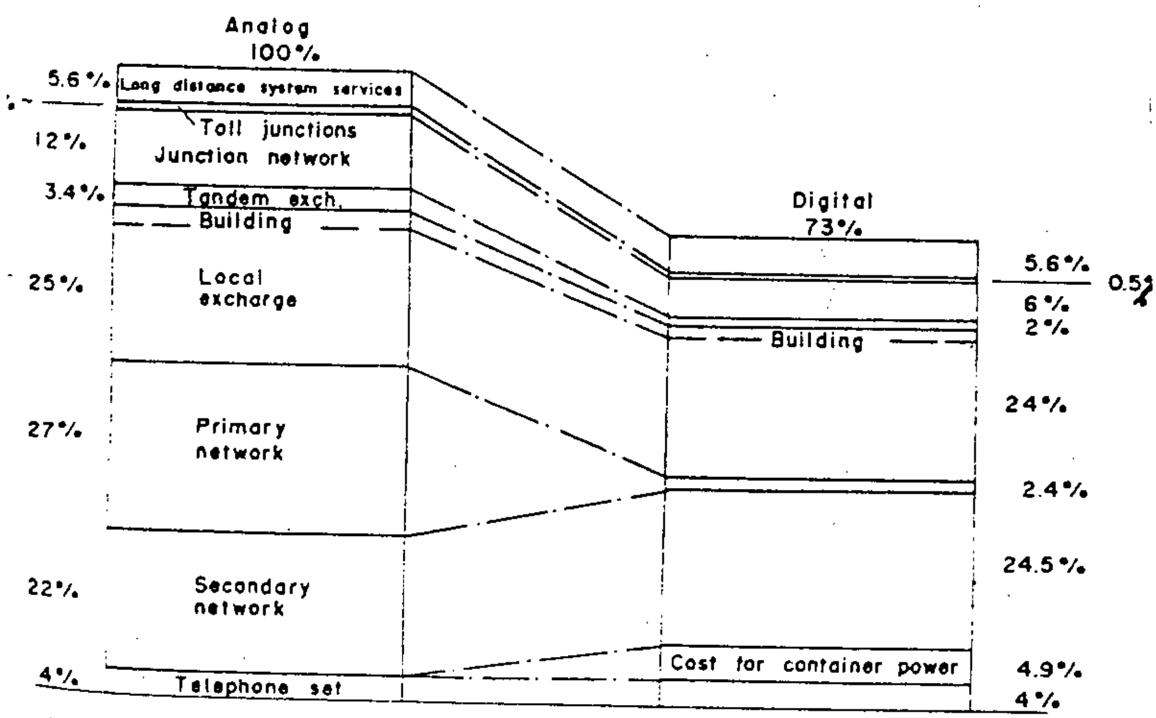


Fig.4.10 *Comparison of Analog and Digital Network Cost Distributions*

4.6 Discussion:

In this chapter a detailed study has been made on PCM Principles. trans ission quality of digital signal, planning consideration, of PCM system over the existing analouge environment. Calculations were carried out for digital communication based on the measured data observed in chapter 3.

It can be predicted that with the introduction of digital transmission the Present 20% traffic hunt failure due to busy state of line/trunks can be successfully overcome with the availability of higher transmission capacity allocated for handling even severe tele-traffic pressure in future.

From the analysis based on existing cable data the following results are obtained:

i) The existing 0.9mm Polythene cables can be equipped with upto 5 PCM systems without further arrangement or restrictions.

ii) If more than 10 PCM systems are required in one line the separation of sending and receiving directions is recommended. That measn one cable will be used for sendinding direction and another one for receiving direction.

iii) The distance between repeaters is to be calculated by using the known quantity of cable data. The difference between measured and real distance is less than 10 percent. As per cable data given in chapter-3 the ohmic limitation of the different routes is within the acceptable standard for implementation of PCM system.

CHAPTER-5

AVAILABLE DIGITAL TRANSMISSION SYSTEM OF THE WORLD.

5.1 Introduction:

Before going into the details of the commonly available digital transmission systems historical background should be narrated.

Digital communications[23] dates back to the telegraph. Many inventors tried to transmit voice and music by code as in the telegraph before telephone was invented by Alexander Graham Bell. Differing from the telegraph code, Bell's invention handled continuous quantities by changing current according to changes in phonetic pressure. Since then, telephone technology has developed on the basis of using the analog transmission scheme to handle continous quantities, and the past forty years have seen remarkable progress.

In the 1930s digital transmission again began to draw interest because of the difficulties incurred with multiplexed transmission through a single line. In the early days of analog frequency division multiplexing (FDM) transmission, filters were very sophisticated and expensive, and crosstalk between channels was considered an almost unsolvable Problem. Some researchers then began to wonder if multiplex transmission in which channels are successively sampled was not more feasible than conventional FDM multiplex transmission where channels are allocated one by one across a spectrum.

The first research on time division multiplex systems started around 1930 in the Pulse amplitude modulation (PAM) scheme. This scheme, however, had the drawbacks of considerable crosstalk, external noise and distortion.

In 1937, Alec. H. Reeves, of ITT Laboratories found that it was possible to change the combination of pulses (0 and 1) as the input voice signal changed. He eventually developed the concept of PCM(Pulse code modulation) in which each sampled voice signal is coded into a combination of pulses for transmission.

After world war II, the wide band and cost barriers were completely removed by the advent and rapid development of transistors and other semiconductor devices. Researchers began to study the application of PCM system to interurban transmission systems about 1954 when commerical semiconductors began to appear.

5.2 Operational PCM Systems:

There are three types[26] of operational PCM system available in the world:

i) The USA 24 channel system -

ii) The UK 24 channel system.

iii) The Euroupen 30-channel system.

The 30-channel-system is now the CCITT international standard adopted by many countries including UK and other European countries, and sub-continent for regional use, and all countries for international use. The CCITT have agreed to the 24 channel USA (T_1) system as an alternative standard for international communications as is used in the USA, Canada and Japan. The UK 24 channel system, which continues to be used nationally, has now been superseded by the CCITT international and regional 30-channel systems.

5.3 Features of the Transmission System PCM 30:

The Primary transmission system[24] PCM 30 Permits Simultaneous transmission of 32 calls via two symmetrical wire pair of an audio cable. Schematic representation of a PCM transmission is shown in Fig. 5.1 :. For this Purpose 30 channel time slots are Provided. Another two channel time slots serve for conveying synchronizing and signalling information and alarms.

5.3.1 Pulse Frame:

Pulse frame: In analogue system we call a VF channel = 0.3 to 3.4 KHz bandwidth. In digital we call a VF channel = 64 K bits. Because sampling rate is 8 KHz and 1 sample is 8 bit. So total number of bits=8x8 kbits = 64 kbits.

The 32 channel time slots of a tranmission system PCM-30 form per sampling interval, 125 μs of a pulse frame. 30 are for speech, one for Frame alignment.

The pulse frame structure is shown in fig. 5.2 which indicates that:

Each pulse frame is made up of 32 channel time slots each slots comprising 8 bits. Hence a pulse frame comprises 32x8 = 256 bits.

The channel time slot duration is 125 μs / 32

= 3.9 μs.

- The 32 channel time slots of the pulse frame are numbered as follows:

= Pulse frame alignment signal or service word (in alternate frames).

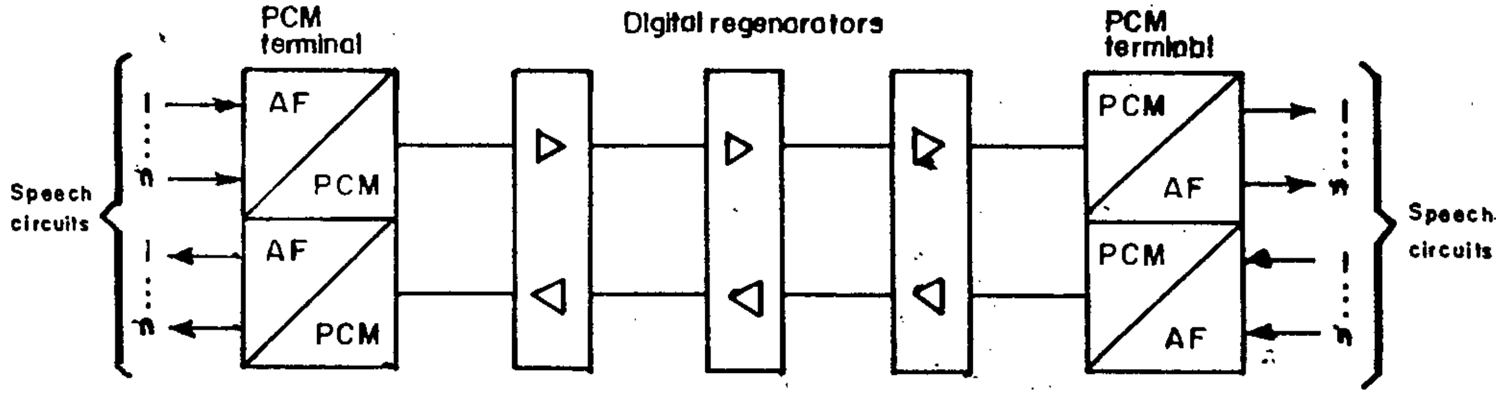


Fig. 5.1 Schematic representation of a PCM transmission system

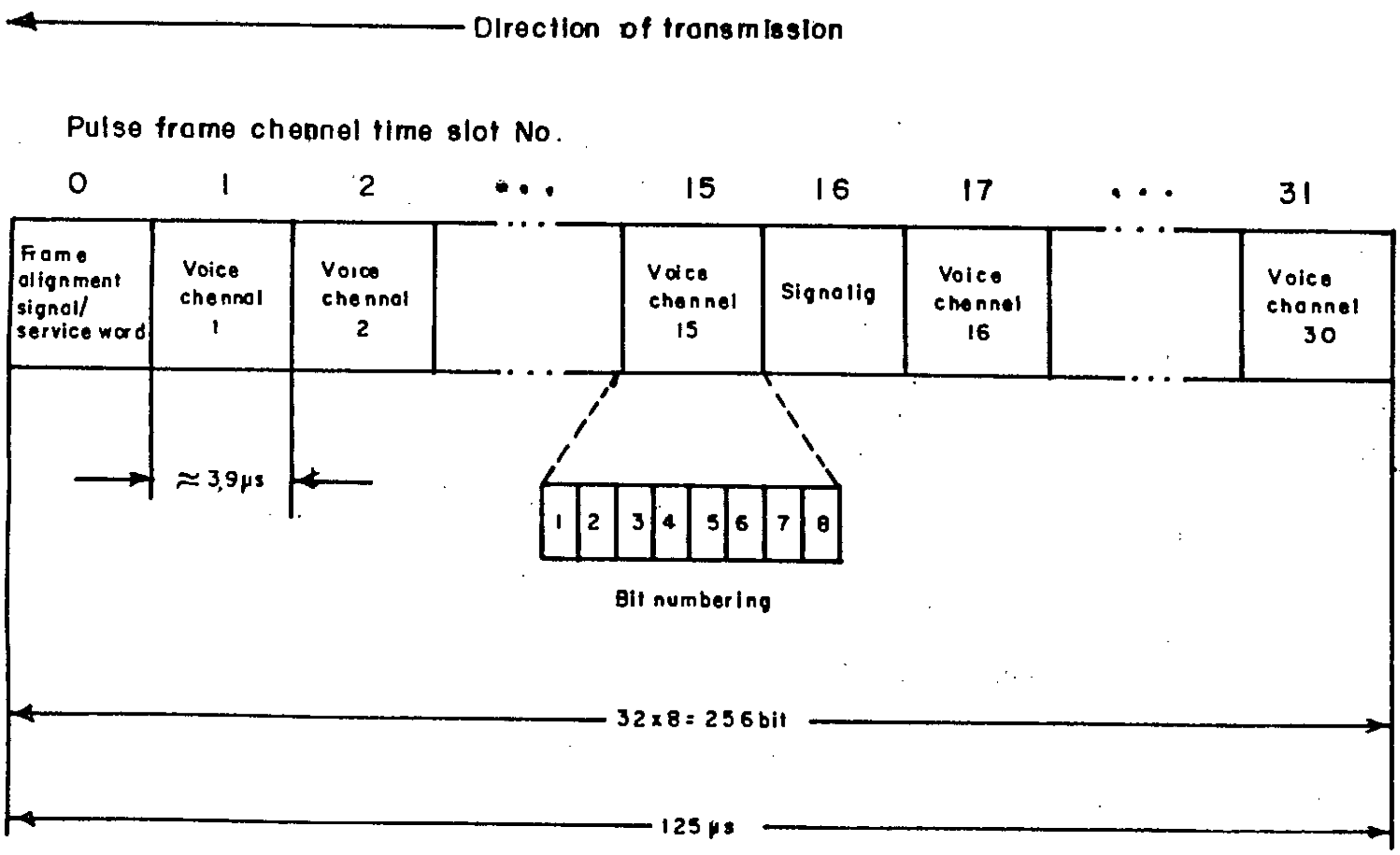


Fig. 5.2 Pulse frame structure of the transmission system PCM 30

1 .. .. .. 15 and)
) Speech channels.
17 .. .. .. 31)

16 Signalling channel.

— The repetition rate of the pulse frame is 8000/s.(sampling frequency 8000Hz).

Bit rate:

— Eight bit encoding is used giving 256 levels to represent the speech samples. Each digit thus occupies 3.9/8 = 0.488 µs made up o of an 0.244 µs pulse and 0.244 µs seperation.

— The bit rate per channel time slot is obtained by multiplying the number of bits per chnnel time slot with the repetition rate: 8 bits x 8000/s = 64 kbits/s.

— The bit rate of a PCM 30 transmission system having 32 channel time slots is 32 x 64 kbits/s = 2048 kbits/s= 2.048 Mbits/s.

— The Permissible tolerance of the PCM 30 transmission system bit rate is $\pm$ 50 bits/s, referred to 10^6 (5×10^{-5}).

5.3.2 Multiframe:

Multiframe: Duration of each time frame is 125 µSec. 16 numbers of time frame together forms a multiframe. Duration of a multiframe is 125 µs x 16 = 2 ms.

5.3.3 Signalling:

Signalling[25] (e.g. answer, clearback and dial signals) is provided for in Time slot 16.

Two types of signalling are available for PCM 30 transmission systems.

a) Channel-associated signalling for 30 speech circuits

b) Signalling via common channel with 64 kbits/sec.

In case of channel-associated signalling (Table 5.1) time slot (TS) 16 is subdivided so that particular bits are available for each of 30 Telephone chnnels. For this reason, 16 pulse frames are combined to form a multiframe. A bunched multiframe alignment signal in channel Time slot 16 of pulse frame 0 is transmitted at the start of the multiframe. The bit pattern of this bunched multiframe alignment signal is "0000". Each of the channel time slots 16 in a multiframe are divided into two groups of 4 bits (a, b, c, d). One of these 4 bit goups in the multiframe is allocated to each of the 30 telephone channels for signalling.

Pulse frame nos.	Bits in channel time slots 16 a	b	c	d	a	b	c	d
0	0	0	0	0	X	Y	X	X
1	Telephone channel			1	Telephone channel			16
2	Telephone channel			2	Telephone channel			17
3	Telephone channel			3	Telephone channel			18
4	Telephone channel			4	Telephone channel			19
5	Telephone channel			5	Telephone channel			20
6	Telephone channel			6	Telephone channel			21
7	Telephone channel			7	Telephone channel			22
8	Telephone channel			8	Telephone channel			23
9	Telephone channel			9	Telephone channel			24
10	Telephone channel			10	Telephone channel			25
11	Telephone channel			11	Telephone channel			26
12	Telephone channel			12	Telephone channel			27
13	Telephone channel			13	Telephone channel			28
14	Telephone channel			14	Telephone channel			29
15	Telephone channel			15	Telephone channel			30

Table 5.1 : Allocation of the bits in the channel time slots 16 of a PCM30 multiframe to the telephone channels for channel-associated signalling.

0000 = bunched multiframe alignment signal.

X = reserve bit

Y = bit for signalling failure of multiframe alignment.

Thus the signalling rate per channel is 4 bits[26] per multiframe i.e. $\frac{4 \times 8k}{16}$ = 2kbits/sec. This is in fact a very generous signalling capacity which is permanently associated with a speech channel even when there is no signalling taking place. The use of the 4 bits for signalling is shown in the following table 5.2. Clearly the use of the 2kbits/sec signalling capacity is very poor from an information theory point of view but it does have the Practical merit of being simple and cheep to install in a PCM multiplexer. In practice the PCM signalling units contain the hybrid transformer which performs the two-to-four-wire conversion (if required) and incorporates the pre-sampling low pass filter. Signalling information is extructed from the co-direction and after signal coding (Table 5.2) sent in TS 16 during the appropriate time in the multiframe.

Table 5.2: Typical Allocation of Signalling codes:

Digits 1-4 or 5-8 appearing serially in TS 16. a	b	c	d	Signalling condition inforward direction.	Signalling condition in backward direction
1	1	1	1	Circuit idle	Circuit busy
0	0	1	1	Circuit seized	Called Sub-answer.
1	0	1	1	Dial Break	--
0	1	1	1	--	Circuit Free
0	0	0	1	Trunk offer	Manual held.
1	0	0	1	-	Coin fee check.
1	1	0	1	Disconnection$^+$	Disconnection$^+$
0	1	0	1	Earth$^+$	Earth$^+$

'+' For use in signalling units that extend AC type signalling by means of phantom circuits derived from transformers associated with each PCM signalling unit.

If timeslot 16 ($\hat{=}$ 64 kbits/sec) is not to be used for channel-associated signalling it can be used for transmitting other digital signals, e.g. for common channel signalling (CCITT No.6, 7) or for data transmission.

5.3.4 Frame Alignment:

Time slot 0 is employed for carrying the frame alignment Pattern and alarm signal. The use of the available 8 bits is shown bel w:

Y 0 0 0 1 0 11	Y 1 Ø X X X X
Alternate frames: frame alignment word.	Alternate frames Not Word

where X : digits not allocated to any particular function and set to 1.

Y : Reserv ed for international use (normally set to 1)

Ø : Digits normally 0 but changed to 1 when loss of frame alignment occurs and/or system-fail alarm occurs.

The spare capacity in the 'Not word' alternate frames may be for network synchronisation signalling.

5.4 24 Channel System(T_1)

This system is used in America, Canada and Japan.

This system has a 125 µs frame with 24 time slots each 8 bits allocated to 24 speech channels. Speech normally is encoded into 8 bits using the µ-law compandor. In Every sixth frame the least significant bit of each channel is used for channel associated signalling giving a 1.667 KHz rate. The frame alignment pattern is carried by a sign bit at the beginning of the frame. Thus the frame contains 1+ 24x8 = 193 bits, the gross system bit rate being 1.544 Mbit/sec. The line code is ADI/AMI giving a line fundamental frequency of 772 KHz.

5.5 UK 24 Channel system:

The 125 μs frame[26] contains 24 time slots each 8 bits allocated to 24 speech is encoded into 7 bits using an A Law compander, occupying bits of 2-8 in time slot. Channel associated signalling is provided by use of bit 1 of each channel every other frame giving a signalling range of 4 KHz. The frame alignment pattern is conveyed by bit 1 of time slots 9 to 24 in (non-signalling)alternate frames. Thus the frame contains 24x 8 = 192 bits, the gross system bit being 1.536 Mbits/sec. The line code is ADI/AMI giving a fundamental line frequency of 768 KHz.

5.6 Comparison of PCM Systems:

Table 5.3 displays the differing parameters for the three PCM systems in operation. The UK 24 channel system is now superceded by the CCITT international and regional 30 channel system. The USA 24-channel system is the alternate CCITT system for regional use only.

Table 5-3 :Comparison of Commercially available PCM Systems:

PARAMETERS	SYSTEM CCITT	24 CH. USA (T1)	24 Channel UK System.
No.of speech channels	30	24	24
No. of 8 bit time slots.	32	24	24
No. of encoded bits forspeech.	8	8*	7
Encoding law	A	μ	A
Signalling	Bunched in TS 16, 4 bits per channel every 16 frames	1 bit per channel every 6 frame.	1 bit per channel every other frame.
Frame alignment pattern.	7 bits bunched in TSO	1 bit per frame	16 bit distributed between TS 9-24 alternate frames.
Bits per frame	256	193	192
Bit rate	2.048 Mbit/s.	1.544 Mbit/s	1.536 Mbits/s.
Line code.	HDB3	ADI/AMI	ADI/AMI.

Note: *7 when channel associated signal bit is used in every 6th frame.

5.7 Higher Order Digital Systems:

The PCM Systems may be further Time Division Multiplexed to higher capacity digital line systems. The international Telegraph and Telephone Consultative Committee (CCITT) has during recent years agreed on a transmission hierarchy on which many national PCM networks are now based.

Two main hierarchies[22] existing within the world today (Fig.5.3, 5.4) are those based on a time division multiplexer that a word interleaves 30 separate speech channels (European, African, South American Scheme), and those that a word interleaves 24 separate speech channels (North American, canadian, Japanese, and the earlier British system). The functions performed by those equipments are identical Only the frequencies involved are different due to the greater bandwidth requirement for accomodating 30 channels compared to 24.

PCM 30 hierarchy :

First order hierarchy;

The binary digit information rate, or bit rate, at the output of the 30-channel multiplex is 2.048 Mbit/s. This bit rate is obtained by interleaving 32 words, or time slots, before the cycle is repeated. Thirty of these words contain the amplitude values from the 30-channel voice signals, while the remaining two time slots contain digits that are used for the synchronization of the demultiplexer to the multiplexer, and signalling information. The transmission medium required for this first order level are symmetrical cable, Radiolink, glass fibre.

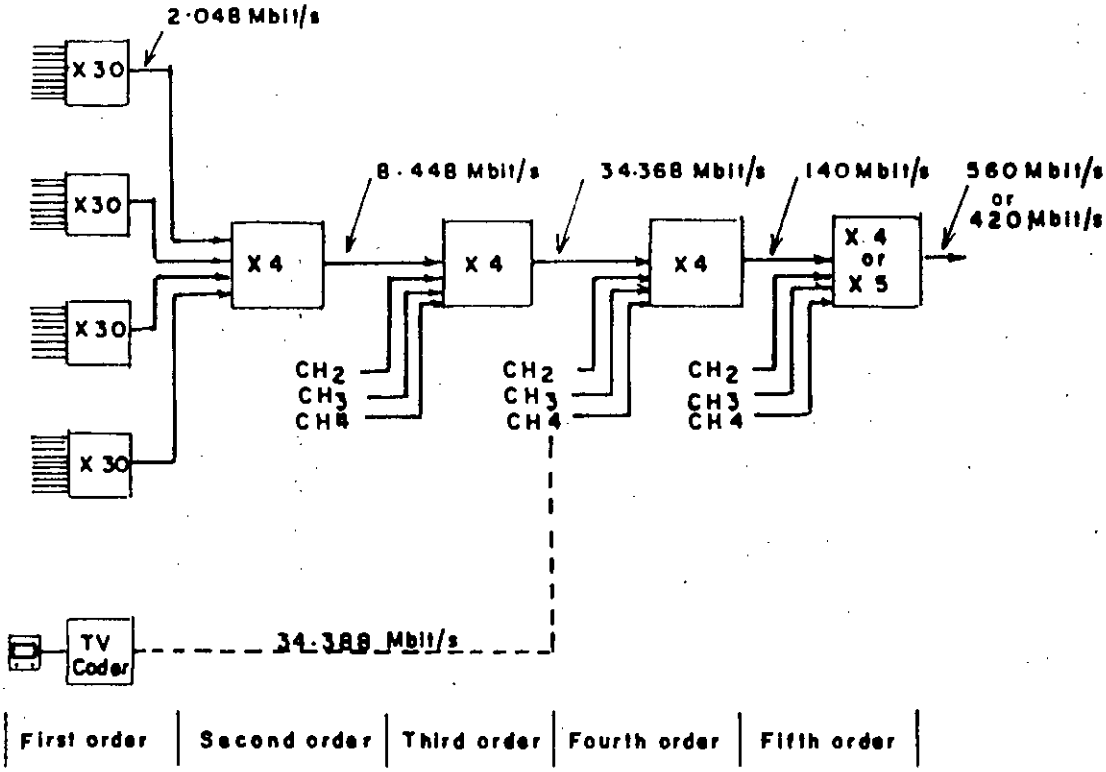


FIG. 5.3 THE PCM 30 HIERARCHY

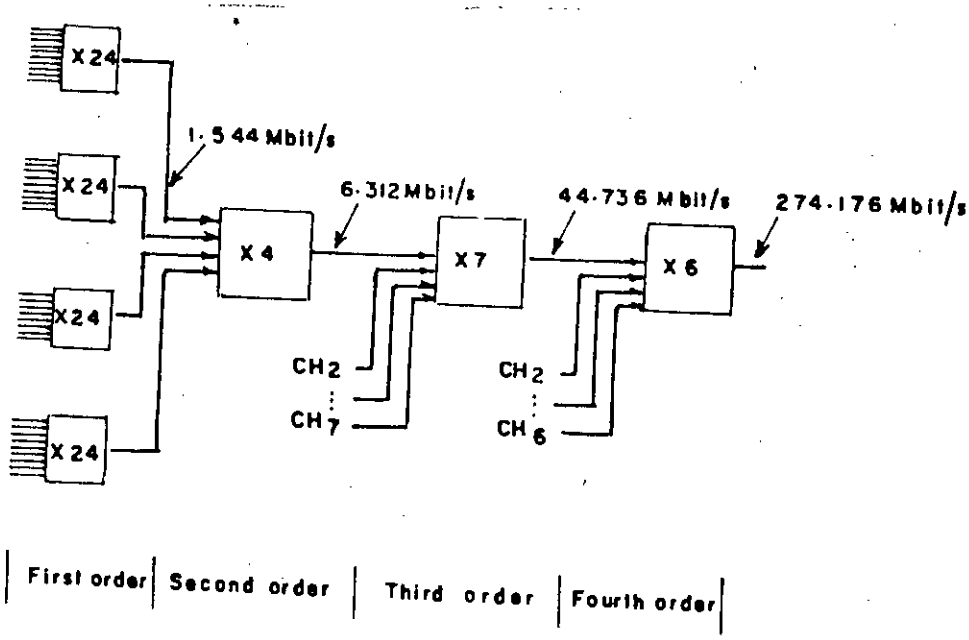


FIG. 5.4 THE PCM 24 HIERARCHY

Second-order hierarchy;

The Second-order equipment contains a multiplex that combines four signals each of bit rate 2.048 Mbit/s into a signal of rate 8.448 Mbit/s. This multiplex contains no encoder and performs the required operation by simply bit interleaving the four input signals. The channel capacity for this Second-order is 120. Special symmetrical cable, co-axial cable and Radio link are the transmission medium for second order equipment.

Third order hierarchy:

Four second order multiplexes are time division multiplexed to form the Third order multiplex of 480 channel at 34.368 Mbits/sec. Transmission medium used for this are special symmetrical cable, co-axial cable of small dia, Radio link, glass fibres.

Fourth order hierarchy:

Four third order multiplexes are time division multiplexed to form the fourth order multiplex of 1960-channel at 139.264 Mbits/s (140 Mbits/s.) Transmission medium required for this types are co-axial cable of small dia, Co-axial cable of large dia, Radio link and glass fibre.

Fifth order hierarchy:

Four fourth order multiplexes are time division multiplexed to form fifth order multiplex of 7840 channel at 560 Mbits/s. The fifth order still lacks international agreement.

24-Channel hierarchical Structure:

The equivalent bit rates for the 24-channel hierarchical structure (Fig. 5.4) as used in North America, Canada and Japan are:

Order	No. of Telephone Channels.	Bit rate
First order.	24	1.544 Mbit/Sec.
Second order	96	6.312 Mbit/Sec.
Third order	672	44.736 Mbits/Sec.
Fourth order	4032	274.176 Mbit/Sec.

5.8 Discussion

In this chapter a comparative study of the characteristics of commercially available PCM systems was carried out. In accordance with the CCITT recommendations PCM 30 has got the optimum properties and is getting wide applications.

CHAPTER-6

GENERAL DISCUSSION

AND

CONCLUSION

In the preceeding chapters a study has been carried out on the feasibility of digital transmission and reception of signal over the existing analogue environment of telecommunication in Bangladesh. The analogue environment, its advantages and disadvantages were studied on the basis of data obtained from major communication routes. For planning of digital system it becomes necessary to study the call handing capacity, problems with multiplexing of analogue signals and possibility of using the existing cables in different routes. These parameters were discussed in chapter-3 after an extensive survey of the existing network made in chapter-2.

It is found that the amount of matured telephone call depends apparently on getting of the Line/Trunk. The Trunk/Line capacity can be increased by using PCM system over existing cables. With the introduction of PCM into the existing system Trunk/line hunting can be increased and the rate of matured telephone calls will greatly increase.

Different problems regarding the multiplexing of the analog signals are studied and it was found that it is affected by inherent thermal and shot noise, crosstalk due to improper filtering and intermodulation noise due to nonlinearities. It is also found that multiplexing units like FDM of analog signals are costlier than digital TDM systems for a given channel capacity.

The measured data of different parameters of the cables (shown in chapter 3) reflects the conditionof cables. It is found that the condition of cables from sher-e-Banglanagar to Mirpur and from Central to Narayanganj is not satisfactory. The insulation resistance in most cases is very low.

In chapter 4 it was analysed that about 20% traffic hunt failure was caused due to busy state of line/Trunks. With the introduction of digital transmission (PCM) this can be successfully overcome.

From the analysis of the existing cable data in chapter 4 the following recommendations can be made in favour of digital transmission (PCM).

i) The existing 0.9mm polythelene cables can be equipped with upto 5 PCM systems in same cable without further arrangement of the cables.

ii) If more than 10 PCM systems are required in one line the separation of sending and receiving direction is recommended. That means one cable will be used for sending direction and another one for receiving direction.

iii) The distance between repeaters is to be calculated by using the known quantity of cable data. The difference between measured and real distance should not be les than 10 percent.

It is also found from the cable data in chapter-3 that the ohmic limitation of the different routes (except sher-e-Banglanagar to Mirpur and Central to Narayanganj) is within the acceptable standard for implementation of PCM system.

It has been pointed out that any communication route can be designed for digital transmission system if the probabilistic traffic data and cable specifications or medium characteristics (for wireless) are given. With the availability of various commercial PCM products as discussed in chapter-5, there remains the scope of consultative works to suggest BT&T

Board for selecting the appropriate equipment. "PCM via optical fibre" is becoming promizing field of digital transmission. Investigation in this line will be very helpful in the future development works on the telecommunication system of Bangladesh.

APPENDIX-A

Measurement of Cable Characteristics

1. Measurement of Insulation Resistance, Loop resistance and capacitance.

Specification of the measuring equipment is as follows:

KABEL MESS KOFFER - KMK-VI

Cable test set - LN-903-013-01

F & G, WEST GERMANY.

Measurements have been taken from main Distribution Frame(MDF) of one Telephone exchange to main Distribution Frame (MDF) of another exchange in accordance with the connection diagram shown in Fig. A-1.

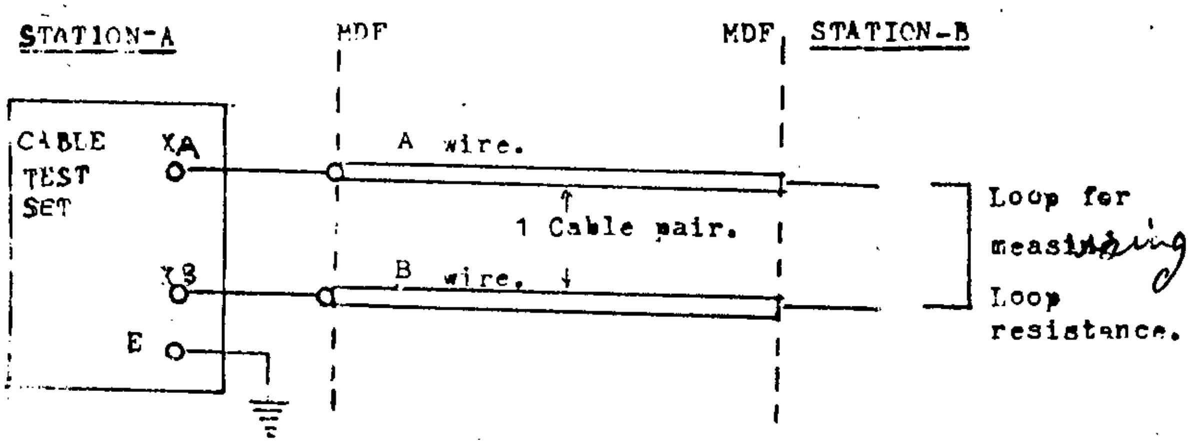


Fig. A-1. Measuring Circuit for obtaining cable parameters.

Measurements: i) Insulation Resistance between

a) A wire to earth and B

b) B wire to earth and A

ii) Loop resistance.

Resistance of both A and B wire in ohm/km.

iii) Capacitance.

Capacitance between A and B wire (nF/km)

The following measurements were taken by the above mentioned equipment.

i) Insulation Resistance between

a) A-wire to earth in Mega-ohms.

b) B-wire to earth in Mega-ohms.

c) A-wire to B-wire in Mega-ohms.

ii) Loop resistance

Resistance of both A and B wires in ohms/Km.

iii) Capacitance

Capacitance between A and B wire in nF/Km.

2. Measurement of cable attenuation at 1 MHz.

The Specification of the Measuring equipments are as follows:

Signal Generator:

Level Generator

PS-12

W&G, West Germany

(Having impedance 124 ohms)

Level meter:

Selective level meter

SPM-12

W&G, West Germany,

(Having impedance of 124 ohms)

As shown in Fig. A-2 the Signal Generator at 1 MHz is connected between A-wire and B-wire of MDF of one exchange and with the help of the level Meter attenuation is measured at MDF of another exchange. The attenuation is measured in dB and dividing it by total length attenuation in dB/Km is obtained.

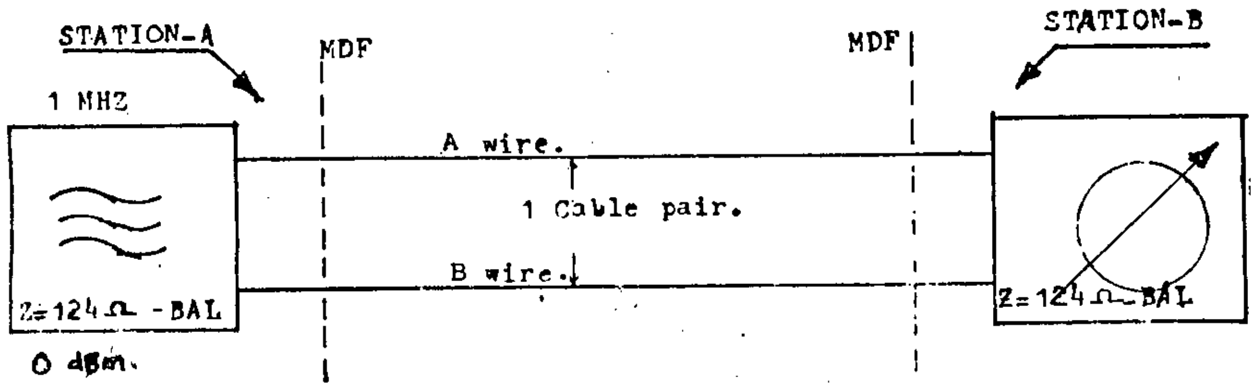


Fig. A-2 Set-up for measurement of cable attenuation.

Appendix-B

Coding laws (non-uniform coding)

Non-uniform coders used in modern PCM systems manage non-uniform quantizing and coding together, according to one of two encoding laws, the A-law or the μ-law. In either case, the encoding law is established on the companding law and code assignment[23].

1. A-law:

According to this law the quantizing level Y_1 for an input level x is determined as

$$Y_1 = \frac{AX}{1 + \log_e A}, \quad 0 \leq X \leq 1/A,$$

$$Y_1 = \frac{1+\log_e(AX)}{1+\log_e A}, \quad 1/A \leq X \leq 1,$$

where A = 87.6

The A-law, recommended by CCITT, Rec. G711, is a 13 segment piecewise linear approximation of the value Y_1, as illustrated in Fig. B-1 Table-A.

2. μ-law (used in America and Japan mainly, and limited to the PCM-24 System)

According to this law the quantizing level Y_2 corresponding to an input level x is determined as

$$Y_2 = \frac{\log_e(1 + \mu X)}{\log_e(1+\mu)}, \quad 0 \leq X \leq 1,$$

where μ = 255.

The μ - law, recommended by CCITT, Rec. G711, is a 15 segment piecewise linear approximation of the value Y_2, as illustrated in Fig.B-1 Table-B.

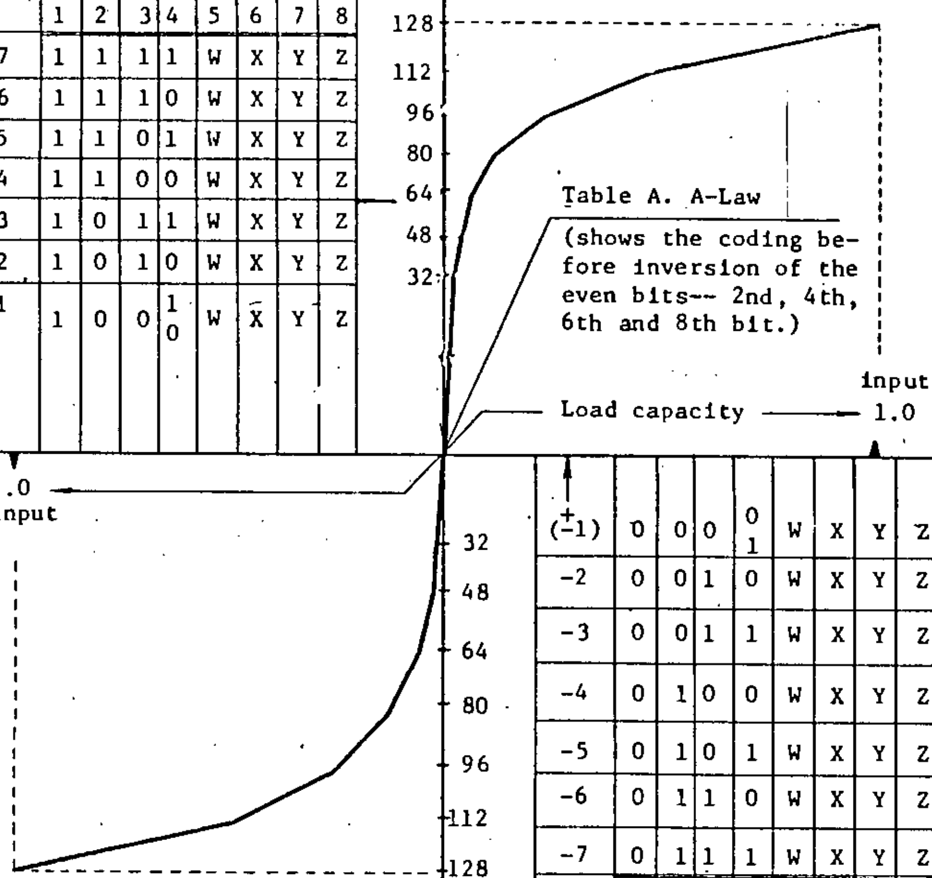
Fig. B-1. Coding laws

Table A. A-Law

(shows the coding before inversion of the even bits-- 2nd, 4th, 6th and 8th bit.)

Segment number	Polarity 1	2	3	4	5	6	7	8
+7	1	1	1	1	W	X	Y	Z
+6	1	1	1	0	W	X	Y	Z
+5	1	1	0	1	W	X	Y	Z
+4	1	1	0	0	W	X	Y	Z
+3	1	0	1	1	W	X	Y	Z
+2	1	0	1	0	W	X	Y	Z
±1	1	0	0	1 0	W	X	Y	Z

(binary expression of quantized samples before code inversion)



Segment number	Polarity 1	2	3	4	5	6	7	8
(±1)	0	0	0	0 1	W	X	Y	Z
-2	0	0	1	0	W	X	Y	Z
-3	0	0	1	1	W	X	Y	Z
-4	0	1	0	0	W	X	Y	Z
-5	0	1	0	1	W	X	Y	Z
-6	0	1	1	0	W	X	Y	Z
-7	0	1	1	1	W	X	Y	Z

(binary expression of quantized samples before code inversion)

Table B. μ-Law

Decision value number	Segment number	Polarity 1	2	3	4	5	6	7	8
128									
112	+8	1	0	0	0	w	x	y	z
	+7	1	0	0	1	w	x	y	z
96	+6	1	0	1	0	w	x	y	z
80	+5	1	0	1	1	w	x	y	z
64	+4	1	1	0	0	w	x	y	z
48	+3	1	1	0	1	w	x	y	z
32	+2	1	1	1	0	w	x	y	z
16	±1	1	1	1	1	w	x	y	z

(binary expression of quantized samples after code inversion)

Decision value number	Segment number	Polarity 1	2	3	4	5	6	7	8
16	(±1)	0	1	1	1	w	x	y	z
32	-2	0	1	1	0	w	x	y	z
48	-3	0	1	0	1	w	x	y	z
64	-4	0	1	0	0	w	x	y	z
80	-5	0	0	1	1	w	x	y	z
96	-6	0	0	1	0	w	x	y	z
112	-7	0	0	0	1	w	x	y	z
128	-8	0	0	0	0	w	x	y	z

(binary expression of quantized samples after code inversion)

The working range of the coder is indicated by the input value normalized to 1.0 in Fig. B-1. This value, 1.0 at maximum, must coincide with the load capacity of the PCM system. The load capacity is +3.14 dBmO in the case of the A-law, and +3.17 dBmO in the case of the μ-law.

The working range can be also expressed with 4096 times of the normalized unit value which corresponds to a half of the minimum decision value in the A-Law at the lowest level segment. In the case of the u-law, the working range is expressed with 8159 units and a unit corresponds to the minimum decision value.

APPENDIX-C.

Calculation of the Number of Repeaters Required

Moghbazar-Central:

Length = 3.7Km.

$$ao = \frac{1}{j} \sum_{j} aoj \quad (j = \text{no. of measurement}).$$

$$ao = \frac{9.7 + 9.7 + 9.8 + 11.0 + 9.4 + 9.5 + 9.4 + 9.5 + 9.5 + 11.1 + 9.9}{11}$$

$$= 9.86 \text{ db/Km at 1 MHz measured.}$$

Maximum Section Loss (aR):

$$aR = aCT - asn - q\,(s.n).$$

acT = 53.5 measured for 0.9 mm cable.

asn = 20 assumed

$$aR = 53.5-20-q(s.n) \quad \text{(one system).}$$

$$= 33.5 \text{ dB (for one system)}$$

$$aR = 53.5-20-10(\text{dB}) \quad \text{(five System).}$$

$$= 23.5 \text{ (for five system).}$$

Length of repeater spacing (LRS)

$$LRS = \frac{aR}{a_o} = \frac{23.5}{9.86} = 2.38 \text{ km.}$$

Length of the starting section (LS)

$$LS = LRS \times 0.5 = 2.38 \times 0.5 = 1.19\text{Km.}$$

No. of Repeater (R) :

$$R = \frac{\text{Total length} - 2\,Ls}{LRS} + 1$$

$$= \frac{3.7 - 2 \times 1.19}{2.38} + 1 = 2.$$

Moghbazar to Shere-e-Banglanagar:

Length = 3.6 Km.

Cable attenuation/Km (40) :

$$ao = \frac{14.5+14.6+14.5+14.6+14.75+14.6+14.6+14.6+14.6+14.8+14.7}{11}$$

$$= 14.6 \text{ db/km.}$$

Maximum Section Loss (aR):

$aR = a_{CT} - asN - q\ (s,n)$:

= 55-20-10 (for five system)

= 25 db.

Length of repeater spacing (LRS) :

$$LRS = \frac{aR}{ao} = \frac{25}{14.6} = 1.71 \text{km.}$$

Length of the starting section (LS):

LS = LRS x 0.5 = 1.71 x 0.5 = 0.86 km.

No. of Repeater(R):

$$R = \frac{3.6-1.71}{1.71} + 1 = 2$$

Shere-e-Banglanagar - Mirpur:

Length = 8.4 km cable : .9mm dia.

$$ao = 0.22 \times \frac{C'}{d}$$

$$= 0.22 \times \frac{35.11}{0.9} = 8.78 \text{ db.} \qquad 9 \text{ db/km.}$$

$aR = a_{CT} - asN - q\ (S.N)$

= 55-20-10

= 25 dB.

$$LRS = \frac{25}{9} = 2.77 \text{ km}$$

$$LS = 2.77 \times .5 = 1.38\text{km}.$$

$$R = \frac{8.4 - 2.77}{2.77} + 1 = 3$$

Moghbazar to Gulshan:

Length = 6.2 km.

Cable attenuation(ao) :

$$ao = \frac{12.0+12.0+11.8+11.8+9.2+9.2+9.2+9.2+9.2+13.9+13.7}{11}$$

$$= 11.01 \text{ db/km}$$

Maximum section loss(aR)

$$aR = a_{CT} - aSN - q\ (S.N).$$

$$= 55-22-10$$

$$= 33\text{dB}.$$

(a_{CT}=55db measured).

aSN = 22 assumed.

Length of Repeater Spacing (LRS):

$$LRS = \frac{aR}{ao} = \frac{33}{11.01} = 2.99\text{km}.$$

$$LS = LRS \times 0.5$$

$$= 2.99 \times 0.5 = 1.49\text{km}.$$

$$\text{No. of Repeaters } (R) = \frac{6.2 - 2.99}{2.99} + 1$$

$$= 2$$

Central-Narayanganj:

Length = 18.3km.

$$ao = 0.22x \frac{C'}{d}$$

$$= 0.22x \frac{34}{0.9}$$

$$= 8,31 \text{ db/km.}$$

$$\simeq 9 \text{ db/km.}$$

Maximum section loss (aR) :

$$aR = aCT - aSN - q\ (S.N)$$

$$= 55-20-10 \text{(for five system).}$$

$$= 25 \text{ db.}$$

Length of the repeater spacing (LRS):

$$LRS = \frac{aR}{ao} = \frac{25}{9} = 2.7\text{km.}$$

Length of the starting section (LS):

$$Ls = LRS \times 0.5 = 2.7 \times 0.5 = 1.35\text{km.}$$

No. of Repeater (R):

$$R = \frac{18.3 - 2.7}{2.7} + 1$$

$$= 5.7 + 1 = 6.7$$

$$= 7 \text{ repeaters.}$$

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