

# TELECOMMUNICATION

AN INDUSTRIAL INTERNSHIP REPORT

*submitted by*

**Aditya Jain**

**(12BEC1016)**

*in partial fulfillment for the award of the degree of*

**BACHELOR OF TECHNOLOGY**

in

**ELECTRONICS AND COMMUNICATION ENGINEERING**



JUNE, 2014



## **School of Electronics Engineering**

### **DECLARATION BY THE CANDIDATE**

I hereby declare that the Industrial Internship Report entitled “**TELECOMMUNICATION**” submitted by me to VIT University, Chennai in partial fulfillment of the requirement for the award of the degree of **Bachelor of Technology in Electronics and Communication Engineering** is a record of bonafide industrial training undertaken by me under the supervision of **Mrs. Aasha Goswami, BSNL, Sagar**. I further declare that the work reported in this report has not been submitted and will not be submitted, either in part or in full, for the award of any other degree or diploma in this institute or any other institute or university.

Chennai

Signature of the Candidate

Date:

# Certificate



## School of Electronics Engineering

### BONAFIDE CERTIFICATE

This is to certify that the Industrial Internship Report entitled “**TELECOMMUNICATION**” submitted by **Aditya Jain (12BEC1016)** to VIT University, Chennai in partial fulfillment of the requirement for the award of the degree of **Bachelor of Technology in Electronics and Communication Engineering** is a record of bonafide internship undertaken by him fulfills the requirements as per the regulations of this institute and in my opinion meets the necessary standards for submission. The contents of this report have not been submitted and will not be submitted either in part or in full, for the award of any other degree or diploma in this institute or any other institute or university.

Dr.Manoj Kumar

Programme Chair (B.Tech ECE)

Date:

---

Examiner (s)	Signature
--------------	-----------

1.	
----	--

2.	
----	--

## **ACKNOWLEDGEMENT**

I am highly thankful to Mrs. Aasha Goswami for providing me the opportunity to undergo the practical training in the TELECOMMUNICATION of BSNL. She guided me from time to time and gave me dynamic ideas and suggestions by which I am able to complete my training successfully.

I am also thankful to manager of various departments whom during the rotations guided me about the works & processing that were being done in their respective departments.

I am also thankful to all those employees of “BSNL” who clarified my doubts & confusion during rotation in the firm. It was a great and satisfying experience having worked in such a professional organization where the process of learning is never ending. I also want to thank all the visible and non-visible hands, which helped me to complete the practical training with great success.

**Aditya Jain**

## TABLE OF CONTENTS

<b>Chapters</b>	<b>Title</b>	<b>Page No.</b>
	Declaration	2
	Certificate	3
	Bonafide Certificate	4
	Acknowledgement	5
	Table of contents	6
	List of Figures	7
	List of Abbreviation	8
	Abstract	9
1.	History of BSNL	10
2.	Types of Telephone Exchange <ul style="list-style-type: none"><li>• C-DOT-Tandom Exchange</li><li>• E-10B-Tandom Exchange</li></ul>	11 11 20
3.	Access network	26
4.	Transmission system	27
5.	GSM	30
6.	Broadband	29
7.	Conclusion	43
8.	Reference	43

## **LIST OF FIGURES**

<b>Fig. No.</b>	<b>Fig. Name</b>	<b>Page No.</b>
1	Hardware Architecture of C-Dot DSS MAX	14
2	Base Module Configuration	16
3	Cellular Structure of GSM Architecture	33
4	GSM Network Architecture	34
5	Core Network Architecture for NIB-II	40
6	Broadband Connectivity in City	41

## **List of Symbols, Abbreviations and Nomenclature**

MDF : Main Distribution Frame

PCM : Pulse Code Modulation  
TAX : Trunk Automatic Exchanges  
GSM : Global System for Mobile Communications  
BTS : Base transceiver station  
BSC : Base station controller  
HLR : Home location register  
VLR : Visitor location registers.  
MSC : Mobile services switching centre  
EIR : Equipment identity registers.  
AuC : Authentication Centre.  
OMC : Operational and Maintenance Centre  
SIM : Subscriber identity module.  
ME : Mobile equipment.  
ADSL : Asymmetric digital subscriber line  
BRAS : broadband remote access server  
MPLS : Multiprotocol Label Switching  
MS : Mobile station

### **Abstract**

I went through a practical Industrial training in BSNL, Sagar (M.P.) on Telecommunications which included the study of Telecommunication architecture, GSM technology and common telephone exchange techniques. It was a great opportunity as well as a nice experience to get to know about the real time working systems of the telecommunication. I had visits to the Base station where I got to look and watch the real time working of the

communications systems. Each aspect of the visit was very fascinating and intriguing as I was allowed to interact with the every component of the entire communication system. It was very good and very different real time experience for me where we faced practical issues on call drops, connection breakage and also had a privilege of watching the solution to them.

### **HISTORY OF BSNL**

Bharat Sanchar Nigam Ltd. formed in October, 2000, is World's 7th largest Telecommunications Company providing comprehensive range of telecom services in India: Wireline, CDMA mobile, GSM Mobile, Internet, Broadband, Carrier service, MPLS-VPN, VSAT, VoIP services, IN Services etc. Presently it is one of the largest & leading public sector unit in India.

BSNL has installed Quality Telecom Network in the country and now focusing on improving it, expanding the network, introducing new telecom services with ICT applications in villages and winning customer's confidence. Today, it has about 46 million line basic telephone capacity, 8 million WLL capacity, 52 Million GSM Capacity, more than 38302 fixed exchanges, 46565 BTS, 3895 Node B ( 3G BTS), 287 Satellite Stations, 614755 R km of OFC Cable, 50430 R km of Microwave Network connecting 602 Districts, 7330 cities/towns and 5.6 Lakhs villages.

BSNL is the only service provider, making focused efforts and planned initiatives to bridge the Rural-Urban Digital Divide ICT sector. In fact there is no telecom operator in the country to beat its reach with its wide network giving services in every nook & corner of country and operates across India except Delhi & Mumbai. Whether it is inaccessible areas of Siachen glacier and North-eastern region of the country. BSNL serves its customers with its wide bouquet of telecom services.

BSNL is numero uno operator of India in all services in its license area. The company offers wide ranging & most transparent tariff schemes designed to suite every customer. BSNL cellular service, CellOne, has 55,140,282 2G cellular customers and 88,493 3G customers as on 30.11.2009. In basic services, BSNL is miles ahead of its rivals, with 35.1 million Basic Phone subscribers i.e. 85 per cent share of the subscriber base and 92 percent share in revenue terms. BSNL has more than 2.5 million WLL subscribers and 2.5 million Internet Customers who access Internet through various modes viz. Dial-up, Leased Line, DIAS, Account Less Internet (CLI). BSNL has been adjudged as the NUMBER ONE ISP in the country. BSNL has set up a world class multi-gigabit, multi-protocol convergent IP infrastructure that provides convergent services like voice, data and video through the same Backbone and Broadband Access Network. At present there are 0.6 million DataOne broadband customers. The company has vast experience in Planning, Installation, network integration and Maintenance of Switching & Transmission Networks and also has a world class ISO 9000 certified Telecom Training Institute. Scaling new heights of success, the present turnover of BSNL is more than Rs.351, 820 million (US \$ 8 billion) with net profit to the tune of Rs.99,390 million (US \$ 2.26 billion) for last financial year. The infrastructure asset on telephone alone is worth about Rs.630, 000 million (US \$ 14.37 billion).

The turnover, nationwide coverage, reach, comprehensive range of telecom services and the desire to excel has made BSNL the No. 1 Telecom Company of India.

## **TYPES OF TELEPHONE EXCHANGE**

### **C-DOT TANDOM EXCHANGE**

## **BRIEF HISTORY : -**

The Center for Development of Telematics (C-DOT) is the telecom technology development center of the government , It was established in August 1984 as an autonomous body. It was vested with full authority and total flexibility to develop state-of-the-art telecommunication technology to meet the needs of the Indian telecommunication network. The key objective was to build a center for excellence in the area of telecom technology .

## **ORGANIZATION : -**

The management of C-DOT has a three-tier structure: **-The governing Council:** provides policy guidelines and approves the annual budget of the center. **The Steering Committee:** has the role of reviewing and monitoring the performance of the center.

**The Project Board:** is responsible for the implementation of C-DOT's project and the day-to-day function of the center.

## **Architecture**

C-DOT switches have distributed architecture .a base moduke (BM) has 512 port which can provide non-blocking connectivity and can accept concentrated subscriber lines. The BM, capable of serving upto 2000 lines or 512 trunks, is used as the basic building block. Each BM is the housed in a single Cabinete. By interconnecting upto 35 such BMs through a central module (CM),the switch can support upto 40,000 lines. The system can support upto 4:1 concentration. Lines and trunks can be intermixed in the same BM.

The front-end of the system consists of an input output processor(IOP) connected to the administration processor(IOP) connected to the administration processor (AP). Each System consist of two IOPs working in duplex mode .IOP, based on Motorola 68040 and supporting UNIX environment, provides interface for the main machine communication, system initialization support ,operator features etc

## **OBJECTIVES: -**

- Work on telecom technology products and services.
- Provide solutions for current and future requirements of telecommunication and converged networks including those required for rural application

- Provide market orientation to R & D activities and sustain C-DOT as center of excellence
- Build partnerships and joint alliances with industry , solution providers, telcos and other development organizations to offer cost effective solution .
- Support telcos and service providers in the introduction of new technologies , features and services by optimal utilization of installed network.

### **MANPOWER : -**

- Electronic Design automation (EDA) Tools for hardware and ASIC Design
- Case Tools for Development and testing of software
- Captive labs
- Computing center
- Pilot production plant
- Existing manpower –907
- Planned Manpower – 963

The Center has state-of-the-art development environment comprising client/ server network of RISC workstation, latest software development tools and very mature and effective development and support methodologies, Extensive use is made of case tools, object-oriented methodologies, software metric etc.The jobs have the latest test and measurement instrument, microprocessor development system and prototyping facilities.

### **ACHIEVEMENTS : -**

- C-DOT Technology based system from 200 lines to 40,000 lines capacity in operation.
- More than 30,000 C-DOT Exchange totaling approximately 25 million telephone lines installed and operational in field.
- Deployed telecom equipment value of Rs.7500 crore.
- Significant technology transfer and royalty earnings.
- Technology development with low capital investment.
- Wide portfolio technologies, products and solution.
- Created large reservoir of technical manpower in telecom

- Established a technology transfer process for production by multiple manufacturers.

## **The C-DOT DSS FAMILY**

C-DOT DSS MAX is a universal digital switch can be configured for different application as local, transit or integrated local and transit switch. High traffic or capacity of 40000 lines as local exchange or 15000 trunks as Trunk automatic exchange. The design of C-DOT DSS MAX has seen by a family concept because of it's advantages like standardized components, commonality in hardware, field hardware that used minimum number of cards, standard cards, racks, frames, cabinets and distribution frames are used which facilitated flexible system growth that make C-DOR DSS MAX easy to maintain and highly reliable.

## **FLEXIBLE ARCHITECTUR**

C-DOT DSS is a modular and flexible digital switching system which provides economical means of serving metropolitan, urban and rural environments. It include all important feature and compulsory services, required by the user with option of up gradation to add new feature and services in future. The architecture for the C-DOT DSS is such that it is possible to upgrade a working C-DOT Single Base Module.(SBM) or Multi Base Module (MBM)exchange to provide Integrated Services Digital Network (ISDN) service by adding minimum addition hardware modules while continue to having existing hardware units. Another factor of architecture Remote Switching Unit(RSU). Is support ISDN. This RSU provides switching facility locally even in case of failure of the communication path to the parent exchange.The resources, which depend upon the number of terminal, are provided within the basic growth unit the Base Module. Base Processors are provided for handling call processing locally. In a small system application, these processors independently support call processing, exchange operation and maintenance function.

## **ARCHITECTURE OF C-DOT DSS MAX**

C-DOT DSS MAX exchanges can be configured using four basic modules.

1. Base Module

2. Central Module
3. Administrative Module
4. Input Output Module

### HARDWARE ARCHITECTURE C –DOT DSS MAX

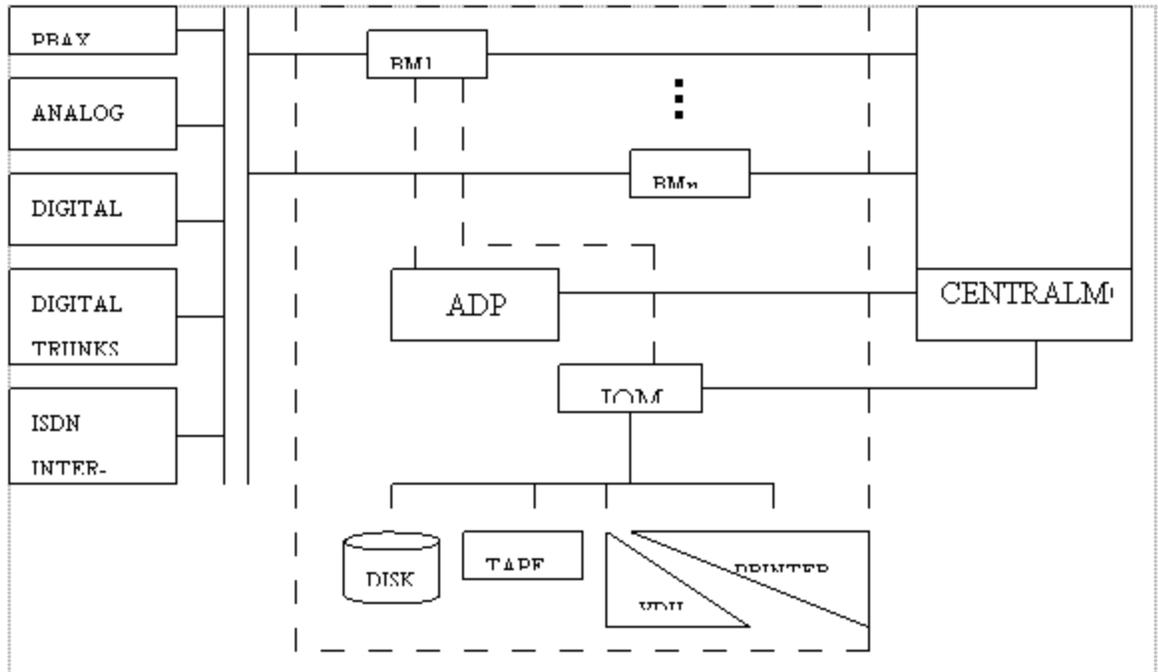


Fig 1: Hardware Architecture of C-Dot DSS MAX

C-DOT MAX exchange can be configured using four basic modules:-

1. BASE MODULE (BM).
2. CENTRAL MODULE (CM) C.
3. ADMINISTRATION STRATIVE MODULE (AM).
4. INPUT OUTPUT MODULE (IOM&IOP).

(a) *BASE MODULE (BM):* -

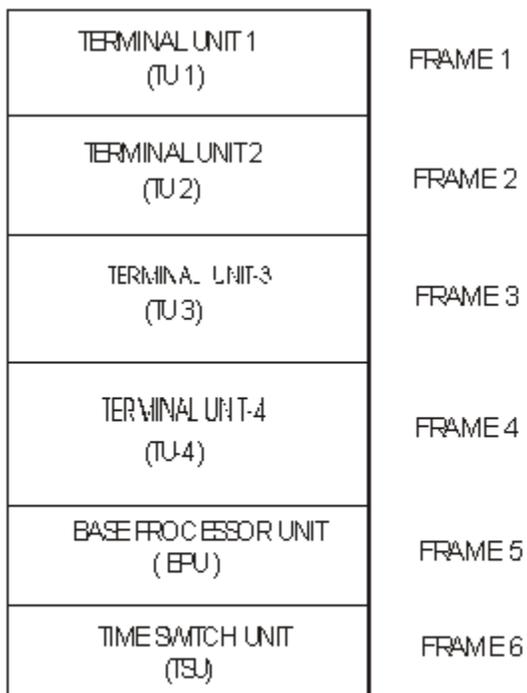
The Base Module is the basic growth unit of the system. It interfaces the external world to the switch. The interfaces may be subscriber lines, Along and digital trunks. Each Base Module can interface up to 2024 terminations. The number of Base Modules directly corresponds to the exchange size. It carries out majority of call processing function and in a small exchange application, it also carries out operation and maintenance function with the help of Input-Output Module.

The Basic functions of a base modules are:-

1. Analog to digital conversion of all signals on analog lines and trunks.
2. Interface to digital trunks and digital subscriber.
3. Switching the calls between terminals connected to the same Base Module.
4. Communication with the AM via the CM for administrative(i.e. Call processing ) functions.
5. Provision of special circuits for call processing support e.g. Digital
6. Tones, announcement, MF/DTMF Senders/receivers.
7. 6. Provision for local switching Unit(RSU) as well as in case of Single Base Module Exchange(SBM\_RAX).

There are two types of Base Modules:-

1. Single Base Modules(SBM)
2. Multi Base Module (MBM)



BASE MODULE(BM) CONFIGURATION

Fig 2: Base Module Configuration

In SBM exchange configuration, the Base Module acts as an independent switching BM directly interface with the Input Output Module for bulk data storage, operations and maintenance function. Clock and synchronization is provided by a source within the BM. It is a very useful application for small urban and rural environments. The Base cabinet houses total 6 frames:-

- Terminal Unit (TU, Top 4 Frames) system and provides connection to 1500 lines and 128 trunks. In such a configuration ,the
- Base Processor Unit ( BPU,5<sup>th</sup> frame)
- Time switch unit (TSU)

There are following four terminals units:-

### **1. ANALOG TERMINAL UNIT (ATU):-**

The Analog Terminals Unit (ATU) is used for interfacing 128 analog termination which may be lines or trunks and providing special circuits as conference announcements and terminal tester. It consists of terminal cards, which may be a combination of Analog Subscriber Line Cards, Analog Trunk card & some Special Service Cards.

#### **(a) Analog Subscriber Ling Cards :-**

Two variants of subscriber line cards as LCC(Line Circuit Card) or CCM(Coin Collection Monitoring) with interfaces upto 8 subscribers. Analog to digital conversion is done by per channel CODEC according to A-Law of Pulse Code Modulation so we can say that it for the subscriber connected for subscriber to exchange. A unit has 16 line cards so  $16*8=128$  subscribers. There are 4 unit so  $4*128= 512$  subscribers.4 cards make 1 Terminal Group(TG) so  $TG = 4$ .

#### **(b) Analog Trunks Cards :-**

Analog trunk cards interface analog inter exchange trunks which may be of three types as TWT,EMT & EMF. These interfaces are similar to subscriber Line Cards, with only difference that the interfaces are designed to scan/drive events on the trunks as predefined signaling requirement.

**(c) Signaling Processor Cards : -**

SP Processes the signaling information received from the terminals cards. SP processes the signaling information consists of scan/drive function like original detection, answer detection, digit reception, reversal detection etc. The validated events are reported to Terminal interface controller for further processing.

**(d) Terminal interface controller (TIC) Cards : -**

TIC controls the four terminals group ( TG) of 32 channels and multiplex them to form a duplicated 128 channels, 8 mbps link towards the Time Switch. For Signaling information of 128 channels it communicates with signaling processor to receive/send the signaling event on analog terminations. It also uses to communicate with BPU.

**(e) Special Service Cards : -**

A Terminal unit has some special service cards such as Conference (CNF) cards to provide six party conference. Speech Samples from five parties are Terminal Test Controller (TTC) card is used to test analog terminal interfaces via the test access relays on the terminal cards. Announcement controller card provides 15 announcement on board cast basis.

**(2) DIGITAL TERMINAL UNIT(DTU) : -**

Digital terminal unit is used to interface digital trunks, i.e. used between the exchanges. one set of Digital Trunks Synchronization (DTS) Card along with the Digital Trunk Controller(DTC) card is used to provide one E-1 interface of 2mbps.Each interface occupies one TG of 32 channels and four such interfaces share 4 TGs in a DTU. Here Terminal Unit Controller (TUC) is used of TIC and DSP cards. Out of 32 channels, 30 for voice communication and remaining two for Signaling and Synchronization. In DTU 4 TGs are there so total number of unit are  $4*30 = 120$  units in DTU.

**(3) # 7 or Signaling Unit Module(SUM) : -**

It is used to support SS7 protocol handlers and some call processing function for CCS7 calls.SS7 capability in C\_DOT DSS MAX exchanges is implemented in the form of a SS& Signaling Unit Module (SUM) The sum hardware is packaged into a standard

equipment frame, similar to that of terminal unit. It is a module by itself and contains global resources. It interfaces with the Time Switch via Terminal Unit Controller (TUC) on a 128 channel PCM link operating at 8mbps.

## **CALL PROCESSING**

### *GENERAL CONCEPT*

There are five function steps of call processing including the location of the originating and terminating equipment. These steps are : -

- **Origination** : - Origination begins when the subscriber line goes off hook or incoming trunks seized. It receives the incoming digits, selects the digit analysis tables, and determines the screening information for this call.
- **Digit Analysis**: - It interprets the digits it receives from origination, select a destination for each call, and passes the dialed digits to routing.
- **Routing/Screening**:- Routing uses the destination information from digit analysis and screening information origination to select the terminating trunk group or line.
- **Charging** : - It uses the charging information from routing to expand the charging data into a format usable by call accounting process.
- **Termination** : - The last step in call processing is termination. Termination Processor is different for calls destined for lines and call destined for trunks.

**Trunk termination** : - A trunk member of the trunks group is selected based on a predetermined pattern. After selection the digits are out pulsed to the distant office.

**Line termination** : - The line identified in routing is checked to determine the line has any special features. Ringing is applied to the line if applicable or the special feature is activated.

## **ROLE OF SOFTWARE IN C-DOT DSS**

### **INTRODUCTION**

The main feature of the software architecture of DSS-MAX are as:

- 1) Distributed architecture to make the distributed control architecture
- 2) Layered architecture with loosely coupled modules & well defined message interfaces.
- 3) Use of high level language
- 4) Modular design with each layer providing higher of abstraction
- 5) Time critical processes in assembly language

These features help in to achieve the following objectives:

- Simplicity in design
- Increased reliability due to fault tolerant software
- Flexibility with option of up gradation to add feature & service
- Efficiency and strict time check
- Ease of Maintainability

## **E-10B TANDOM EXCHANGE**

### **ESTABLISHMENT OF E10B**

The first E-10-B system (a training model) was commissioned at ALTCC, Ghaziabad in July 84. The first commercial E-10-B system was setup at Bombay in April 85 supporting 10,000 lines. Also 22,000 lines of digital TAX (E-10-B type) have been installed at 16 stations all over the country. (The first at Agra in Feb. 87) Palaghat (Kerala) unit of ITI manufactures E-10-B TAX equipments. Another ITI factory at MANKAPUR in GONDA (UP) produces annually 500,000 E-10B local lines

### **EVOLUTION OF E-10B SYSTEM**

The predecessor of E-10B is the E-10A system developed in France in early Seventies. Based on the structure of E-10A, a more powerful system with a significantly higher call handling / traffic capacity was developed in early 80's. The first E-10 system was commissioned at BREST in FRANCE. The system has many versions. The INDIAN version is the 384 PCM versions and can handle max traffic capacity of 4000 erlangs.

The BHCA is 1,90,000.

## **APPLICATIONS OF THE E-10-B SYSTEM**

### **(a) LOCAL EXCHANGES**

These exchanges terminate local subscriber lines and are connected to other exchanges in the local network. The limit of max traffic handling capacity is 4000 erlangs. Within this, any proportion of subscribers and junctions is possible.

### **(b) LOCAL, TRANSIT, TANDEM EXCHANGES**

E-10B system can be used to carry pure transit traffic. Here, subscribers line providing terminating equipments will not be provided. Only equipments needed for connecting junctions will be provided.

### **(c) TAX**

Here, the system provides for termination of long distances circuits. Digital Tax's has a max capacity of 11,000 lines (o/g and I/g) in 384 versions.

### **(d) LOCAL CUM TRANSIT OR TAX**

The facilities of local and transit can be combined.

## **FACILITIES AND SERVICES TO THE SUBSCRIBER**

1. Call forwarding
2. Short code Dialing
3. Malicious call tracing
4. Conference calls
5. Call waiting
6. Detailed billing
7. Automatic alarm call
8. Barred access
9. Hotline facility
10. Pushbutton telephone

## 11. Last number redial

Because of its modular structure, E-10-B can be expanded to meet its demands and new services can be introduced with modification of software.

### **Features of E10-B system**

The system is based on these features.

- (1) Stored program control
- (2) TDM digital switching
- (3) PCM principals and techniques
- (4) Segregation of switching and management facilities.
- (5) Distributed control using dedicated microprocessors,  
e.g. 8085, ELS-48
- (6) Centralized management for a group of E-10-B exchanges.

#### **Stored Program Control**

Control functions relating to call processing are carried out by execution of program instructions stored in the memory of computers. In electromechanical systems, these functions are hardware based. In E-10B, these are software based.

#### **TDM Digital Switching**

The system switches signals to digital form. Analog signals are converted to TDM multiplexed digital signals, prior to switching.

#### **PCM Principles**

Systems have been developed for 30 channels PCM corresponding to Relevant CCITT recommendations.

## **Segregation b/w switching and management functions**

Switching functions like reception of dialed digits, the storage, the analysis, routing of the call etc. are performed by the control unit in the exchange, which has a decentralized architecture, employing dedicated processors. Functions like subscriber lines and circuit group management, faults and alarm management etc. are done by a separate mini computer, located at a centralized operation and maintenance center (OMC), which is common for a number of E-10-B exchanges. OMC and switching centers are interconnected by PCM links. They could be in the same- premises *or* far apart.

### **Distributed control**

Call handling and call processing functions like scanning of subscriber lines, detection of loop status, reception and storage of digits etc. are distributed over functional units. Dedicated processors like Intel 8085 and dedicated mini computers like ELS-48 handle them.

### **Centralized management for a group of E – 10 B exchanges**

The O&M functions for a group of E-10-B exchanges (upto a maximum of 6 exchanges or '80,000 lines) are carried out by single OMC, which is connected to various exchanges by PCM links.

The E-10-B exchange consists of three blocks:

- a. Connection units
- b. Switching networks
- c. Control units

### **Connection Units**

They act as an interface between external environment (i.e. subscribers and trunks) and central units. Units, which manage the generation and transmission of, digitalized tones and frequencies and dissemination of recorded announcements to subscribers are also called connection units.

### **Switching networks (CX)**

This is a TDM switching network having a three stage time-space-time architecture. It uses four wires switching (4W) for connecting time slots of calling and called parties.

### **Control Units**

The control unit handles telephone call setup, supervision, clear down and charging functions.

#### **Specifications of E-10B:-**

1. Number of switchable PCM links: **384**
2. Processing capacity: **1,90,000** BHCA
3. Traffic handling capacity: **4000** erlangs
4. Subscriber exchange: **45,000** lines & **5000** circuits

#### **SYSTEM FEATURES:-**

1. Time division multiplexing
2. PCM to CCITT standards
3. 2 Mbit/s PCM link
4. 30 telephone channels per PCM link
5. 8 bits per telephone channel
6. Stored program control
7. Dedicated processors for program control
8. Non dedicated processor for operation functions

#### **SUBSCRIBER LINE**

1. Dial or pushbutton VF telephone

2. Maximum loop resistance inclusive of telephone set 2400 ohms
3. Ringing current: 80 V, 25 and 50 Hz

### **ENVIRONMENTAL CONDITIONS**

#### **Exchange:**

1. Ambient temp of air drawn into racks: 18 - 20 deg C
2. Relative humidity: 30 - 70 %

#### **Satellite exchange:**

1. Ambient temp: 5 -35 deg C
2. Relative humidity: 20 - 80 %

#### **DIMENSIONS OF E-10B:-**

1. Rack dimension: height: 2 mt, width 0.75 mt, depth 0.5 mt, Distributive floor loading, less **than** 500 kg/sq. **mt**
2. Floor area: 45,000 lines subscriber exchange = 154 sq. **mt**  
11,000 lines subscriber exchange = 90 sq. **mt**

#### **POWER SUPPLY REQUIREMENTS:-**

1. Exchange and satellite exchange = -48 V
2. OMC: 220 V, 50 Hz
3. Power supply current online: 23 - 60 mA
4. Loop resistance: 1500 to 2400 ohms

### **CALL PROCESSING IN E-IO B EXCHANGE**

The following types of call processing has been described:

- 1) Local call
- 2) I/C call
- 3) O/g call

A call is processed in the following four stages:

- 1) Pre-selection
- 2) Selection
- 3) Connection and charging
- 4) Release

#### **PURPOSE OF MDF AND DDF IN E-10B EXCHANGE**

E-10B switching system is a digital switching system and accepts both analog and digital signals. The signals from subscriber are analog whereas the trunk signals are digital (if coming from analog source, these are converted into digital 30 channel PCM signals and then fed to switch room).

The analog signals are received by CSE and digital signals by URM in E-10B system. For these two types of signals we use two types distribution frames in E-10B. One is called Main Distribution Frame (MDF) and the other one is called Digital Distribution Frame (DDF).

#### **Access network**

An access network is that part of a communications network which connects subscribers to their immediate service provider. It is contrasted with the core network, for example the Network Switching Subsystem in GSM. The access network may be further divided between feeder plant or distribution network, and drop plant or edge network.

#### **Fixed Line Access Network - Telephone**

An access network or outside plant refers to the series of wires, cables and equipment lying between a consumer/business telephone termination point (the point at which a telephone connection reaches the customer) and the local telephone exchange. The local exchange contains banks of automated switching equipment to direct a call or

connection to the consumer. The access network is perhaps one of the oldest assets a telecoms operator owns, and is constantly evolving, growing as new customers are connected, and as new services are offered. This makes the access network one of the most complex networks in the world to maintain and keep track of. In 2007-2008 many telecommunication operators experienced increasing problems maintaining the quality of the records which describe the network. In 2006, according to an independent Yankee Group report, globally operators experience profit leakage in excess of €15 Billion each year. The access network is also perhaps the most valuable asset an operator owns, since this is what physically allows them to offer a service.

Access networks consist largely of pairs of copper wires, each traveling in a direct path between the exchange and the customer. In some instances, these wires may even be aluminum, the use of which was common in the 1960s and 1970s following a massive increase in the cost of copper. As it happened, the price increase was temporary, but the effect of this decision is still felt today because the aluminum wires oxidize and lose their ability to carry large quantities of data. Access is essential to the future profitability of operators who are experiencing massive reductions in revenue from POTS (plain old telephone services), due in part to the opening of historically nationalized companies to competition, and in part to increased use of mobile phones and VOIP (voice over IP) services. Operators now look toward additional services such as xDSL based broadband and IPTV (Internet Protocol Television) to guarantee future profit. The access network is again the main barrier to achieving these profits since operators world wide have accurate records of only 40% to 60% of the network. Without understanding or even knowing the characteristics of these enormous copper spider webs, it is very difficult, and expensive to 'provision' (connect) new customers and assure the data rates required to receive next generation services. Over time, we will see the access networks around the world evolve to include more and more optical fiber technology. Optical fibre already makes up the majority of core networks and will start to creep closer and closer to the customer, until a full transition to 21st Century Networks is achieved, delivering value added services over fiber to the home (FTTH).

***(Access process) Access Network Authentication High-Level Example***

The process of communicating with a network begins with an access attempt, in which one or more users interact with a communications system to enable initiation of user information transfer. An access attempt itself begins with an issuance of an access request by an access originator. An access attempt ends either in successful access or in access failure - an unsuccessful access that results in termination of the attempt in any manner other than initiation of user information transfer between the intended source and destination (sink) within the specified maximum access time. Access failure can be the result of access outage, user blocking, incorrect access, or access denial. Access denial (system blocking) can include: Access failure caused by the issuing of a system

blocking signal by a communications system that does not have a call-originator camp-on feature. Access failure caused by exceeding the maximum access time and nominal system access time fraction during an access attempt. Without an access network, a fixed line telco can not exist, yet this network has been undervalued and under invested for decades. Telcos today need to massively improve their understanding of these networks to remain profitable in the short term and remain in existence in the longer term. British Telecom has produced an interactive presentation introducing the technology and design of access networks

### **Charging for access**

An access charge is a charge made by a local exchange carrier for use of its local exchange facilities for a purpose such as the origination or termination of traffic that is carried to or from a distant exchange by an interexchange carrier

## **Transmission system**

In telecommunications a transmission system is a system that transmits a signal from one place to another. The signal can be an electrical, optical or radio signal. Some transmission systems contain multipliers, which amplify a signal prior to re-transmission, or regenerators, which attempt to reconstruct and re-shape the coded message before re-transmission.

### **Pulse-code modulation**

Pulse-code modulation (PCM) is a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals, then quantized to a series of symbols in a digital (usually binary) code. PCM has been used in digital telephone systems and is also the standard form for digital audio in computers and the compact disc red book format. It is also standard in digital video, for example, using ITU-R BT.601. However, straight PCM is not typically used for video in consumer applications such as DVD or DVR because it requires too high a bit rate (PCM audio is supported by the DVD standard but rarely used). Instead, compressed variants of PCM are normally employed. However, many Blu-ray Disc movies use uncompressed PCM for audio. Very frequently, PCM encoding facilitates digital transmission from one point to another (within a given system, or geographically) in serial form.

## **History of PCM**

In retrospect, PCM, like many other great inventions, appears to be simple and obvious. In the history of electrical communications, the earliest reason for sampling a signal was to interlace samples from different telegraphy sources, and convey them over a single

telegraph cable. Telegraph time-division multiplexing (TDM) was conveyed as early as 1853, by the American inventor M.B. Farmer. The electrical engineer W.M. Miner, in 1903, used an electro-mechanical commutator for time-division multiplex of multiple telegraph signals, and also applied this technology to telephony. He obtained intelligible speech from channels sampled at a rate above 3500–4300 Hz: below this was unsatisfactory. This was TDM, but pulse-amplitude modulation (PAM) rather than PCM. Paul M. Rainey of Western Electric in 1926 patented a facsimile machine using an optical mechanical analog to digital converter. The machine did not go into production. British engineer Alec Reeves, unaware of previous work, conceived the use of PCM for voice communication in 1937 while working for International Telephone and Telegraph in France. He described the theory and advantages, but no practical use resulted. Reeves filed for a French patent in 1938, and his U.S. patent was granted in 1943.

The first transmission of speech by digital techniques was the SIGSALY vocoder encryption equipment used for high-level Allied communications during World War II from 1943. It was not until about the middle of 1943 that the Bell Labs people who designed the SIGSALY system, became aware of the use of PCM binary coding as already proposed by Alec Reeves. PCM in the 1950s used a cathode-ray coding tube with a grid having encoding perforations. As in an oscilloscope, the beam was swept horizontally at the sample rate while the vertical deflection was controlled by the input analog signal, causing the beam to pass through higher or lower portions of the perforated grid. The grid interrupted the beam, producing current variations in binary code. Rather than natural binary, the grid was perforated to produce Gray code lest a sweep along a transition zone produce glitches

### **Managed Leased Line Network (MLLN)**

Leased Line services with flexible access bandwidth. The MLLN is a Managed Leased Line Network system which is proposed to provide Leased line connectivity. With the State-of-the-art technology equipment, MLLN is designed mainly for having effective control and monitor on the leased line so that the down time is very much minimised and the circuit efficiency is increased thus achieving more customer satisfaction. This mainly deals with data circuits ranging from 64 KBPs to 2048 KBPs.

In MLLN network conventional PCM primary MUX and subscriber Modems are replaced by versatile MUX and network terminating units respectively. MLLN mainly consists of Digital Cross Connect (DXC), versatile MUX (V MUX), Network Terminating Units (NTU) and Network Management System (NMS). DXC's and VMUX's are inter connected via optical fibre links with alternate routing facility in case of any route failure. VMUX is in turn connected to NTU's placed at customer premises through 2 wire copper pair. At the top of it NMS is suitably placed at the Central location for effective control & monitor. NTUs are fully managed from NMS. They are

programmable for different data speeds ranging from 64 KBPs (n x64 KBPs: nx1 to 32) depending upon the customer demand, thus having bandwidth control without changing Modem at his premises. NTUs operate on 230 V AC.

## **Features of MLLN**

- Control, Manage the leased line network.
- Bandwidth management as per the customer demand.
- Pro-active maintenance, without waiting for customer to book a complaint.
- Self Diagnostic/software loops to check E1 connectivity to DXC, VMUX/software loops for checking copper pair at NTU point for immediately identifying the faulty section for trouble shooting .
- Alternate routing in case of any route failure.
- Generation of the periodic performance reports for self-analysis/customer.

## **Synchronous Digital Hierarchy**

SDH (Synchronous Digital Hierarchy) is a standard technology for synchronous data transmission on optical media. It is the international equivalent of Synchronous Optical Network. Both technologies provide faster and less expensive network interconnection than traditional PDH (Plesiochronous Digital Hierarchy) equipment. In digital telephone transmission, "synchronous" means the bits from one call are carried within one transmission frame. "Plesiochronous" means "almost (but not) synchronous," or a call that must be extracted from more than one transmission frame. SDH uses the following Synchronous Transport Modules (STM) and rates: STM-1 (155 megabits per second), STM-4 (622 Mbps), STM-16 (2.5 gigabits per second), and STM-64 (10 Gbps).

## **GSM(Global System for Mobile Communications)**

GSM (Global System for Mobile Communications, originally Groupe Spécial Mobile), is a standard set developed by the European Telecommunications Standards Institute (ETSI) to describe technologies for second generation (or "2G") digital cellular networks. Developed as a replacement for first generation analogue cellular networks, the GSM standard originally described a digital, circuit switched network optimized for full duplex voice telephony. The standard was expanded over time to include first circuit switched data transport, then packet data transport via GPRS. Packet data transmission speeds were later increased via EDGE. The GSM standard is succeeded by the third generation (or "3G") UMTS standard developed by the 3GPP. GSM networks will evolve further as they begin to incorporate fourth

generation (or "4G") LTE Advanced standards. "GSM" is a trademark owned by the GSM Association.

The GSM Association estimates that technologies defined in the GSM standard serve 80% of the global mobile market, encompassing more than 1.5 billion people across more than 212 countries and territories, making GSM the most ubiquitous of the many standards for cellular networks.

## Introduction to GSM Networks

The various interface labels are the formal names given to these interfaces. More details about these interfaces are found in GSM TS 03.02 [26].

### **The GSM network consists mainly of the following functional parts:**

- i. **MSC** – the mobile service switching centre (MSC) is the core switching entity in the network. The MSC is connected to the radio access network (RAN); the RAN is formed by the BSCs and BTSs within the Public Land Mobile Network (PLMN). Users of the GSM network are registered with an MSC; all calls to and from the user are controlled by the MSC. A GSM network has one or more MSCs, geographically distributed.
- ii. **VLR** – the visitor location register (VLR) contains subscriber data for subscribers registered in an MSC. Every MSC contains a VLR. Although MSC and VLR are individually addressable, they are always contained in one integrated node.
- iii. **GMSC** – the gateway MSC (GMSC) is the switching entity that controls mobile terminating calls. When a call is established towards a GSM subscriber, a GMSC contacts the HLR of that subscriber, to obtain the address of the MSC where that subscriber is currently registered. That MSC address is used to route the call to that subscriber.
- iv. **HLR** – the home location register (HLR) is the database that contains a subscription record for each subscriber of the network. A GSM subscriber is normally associated with one particular HLR. The HLR is responsible for the sending of subscription data to the VLR (during registration) or GMSC (during mobile terminating call handling).
- v. **CN** – the core network (CN) consists of, amongst other things, MSC(s), GMSC(s) and HLR(s). These entities are the main components for call handling and subscriber management. Other main entities in the CN are the equipment identification register (EIR) and authentication centre(AUC). CAMEL has no interaction with the EIR and AUC; hence EIR and AUC are not further discussed.

- vi. **BSS** – the base station system (BSS) is composed of one or more base station controllers (BSC) and one or more base transceiver stations (BTS). The BTS contains one or more transceivers
- vii. **(TRX)**- The TRX is responsible for radio signal transmission and reception. BTS and BSC are connected through the Abis interface. The BSS is connected to the MSC through the A interface.
- viii. **MS** – the mobile station (MS) is the GSM handset. The structure of the MS will be described in more detail in a next section. A GSM network is a public land mobile network (PLMN). Other types of PLMN are the time division multiple access (TDMA) network or code division multiple access (CDMA) network. GSM uses the following sub-division of the PLMN:
- ix. **Home PLMN (HPLMN)** – the HPLMN is the GSM network that a GSM user is a subscriber of. That implies that GSM user's subscription data resides in the HLR in that PLMN. The HLR may transfer the subscription data to a VLR (during registration in a PLMN) or a GMSC (during mobile terminating call handling). The HPLMN may also contain various service nodes, such as a short message service centre (SMSC), service control point (SCP), etc.
- x. **Visited PLMN (VPLMN)** – the VPLMN is the GSM network where a subscriber is currently registered. The subscriber may be registered in her HPLMN or in another PLMN. In the latter case, the subscriber is outbound roaming (from HPLMN's perspective) and inbound roaming (from VPLMN's perspective). When the subscriber is currently registered in her HPLMN, then the HPLMN is at the same time VPLMN.
- xi. **Interrogating PLMN (IPLMN)** – the IPLMN is the PLMN containing the GMSC that handles mobile terminating (MT) calls. MT calls are always handled by a GMSC in the PLMN, regardless of the origin of the call. For most operators, MT call handling is done by a GMSC in the HPLMN; in that case, the HPLMN is at the same time IPLMN. This implies that calls destined for a GSM subscriber are always routed to the HPLMN of that GSM subscriber. Once the call has arrived in the HPLMN, the HPLMN acts as IPLMN. MT call handling will be described in more detail in subsequent sections. When basic optimal routing (BOR) is applied, the IPLMN is not the same PLMN as the HPLMN. The user of a GSM network is referred to as the served subscriber ; the MSC that is serving that subscriber is known as the serving MSC.
- xii. **Mobile originated call** – the MSC that is handling the call is the serving MSC for this call; the calling subscriber is the served subscriber.
- xiii. **Mobile terminated call** – the GMSC that is handling the call is the serving GMSC for this call; the called subscriber is the served subscriber.

## Signalling in GSM

The various entities in the GSM network are connected to one another through signalling networks. Signalling is used for example, for subscriber mobility, subscriber registration, call establishment, etc. The connections to the various entities are known as 'reference points'. Examples include:

- a. A interface – the connection between MSC and BSC;
- b. Abis interface – the connection between BSC and BTS;
- c. D interface – the connection between MSC and HLR;
- d. Um interface – the radio connection between MS and BTS.

Various signaling protocols are used over the reference points. Some of these protocols for GSM are the following:

- a. mobile application part (MAP) – MAP is used for call control, subscriber registration, short message service, etc.; MAP is used over many of the GSM network interfaces;
- b. base station system application part (BSSAP) – BSSAP is used over the A interface;
- c. direct transfer application part (DTAP) – DTAP is used between MS and MSC; DTAP is carried over the Abis and the A interface. DTAP is specified in GSM TS 04.08 [49];
- d. ISDN user part (ISUP) – ISUP is the protocol for establishing and releasing circuit switched calls. ISUP is also used in landline Integrated Services Digital Network (ISDN). A circuit is the data channel that is established between two users in the network. Within ISDN, the data channel is generally a 64 kbit/s channel. The circuit is used for the transfer of the encoded speech or other data. ISUP is specified in ITU-T Q.763 [137].
- e. When it comes to call establishment, GSM makes a distinction between signaling and payload. Signaling refers to the exchange of information for call set up; payload refers to the data that is transferred within a call, i.e. voice, video, fax etc. For a mobile terminated GSM call, the signaling consists of exchange of MAP messages between GMSC, HLR and visited MSC (VMSC). The payload is transferred by the ISUP connection between GMSC and VMSC. It is a continual aim to optimize the payload transfer through the network, as payload transfer has a direct cost aspect associated with it. Some network services are designed to optimize the payload transfer. One example is optimal routing.

**The network is structured into a number of discrete sections:**

- The **Base Station Subsystem** (the base stations and their controllers).
- The **Network and Switching Subsystem** (the part of the network most similar to a fixed network). This is sometimes also just called the core network.

### GSM as a network of cells

The GSM or any other wireless technology is been divided into cells in order to increase the system capacity and coverage range.

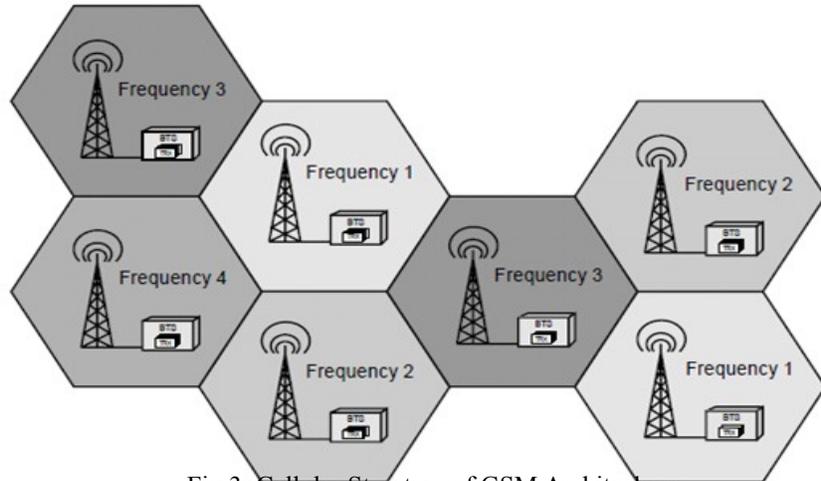


Fig 3: Cellular Structure of GSM Architecture

**GSM Network Architecture**

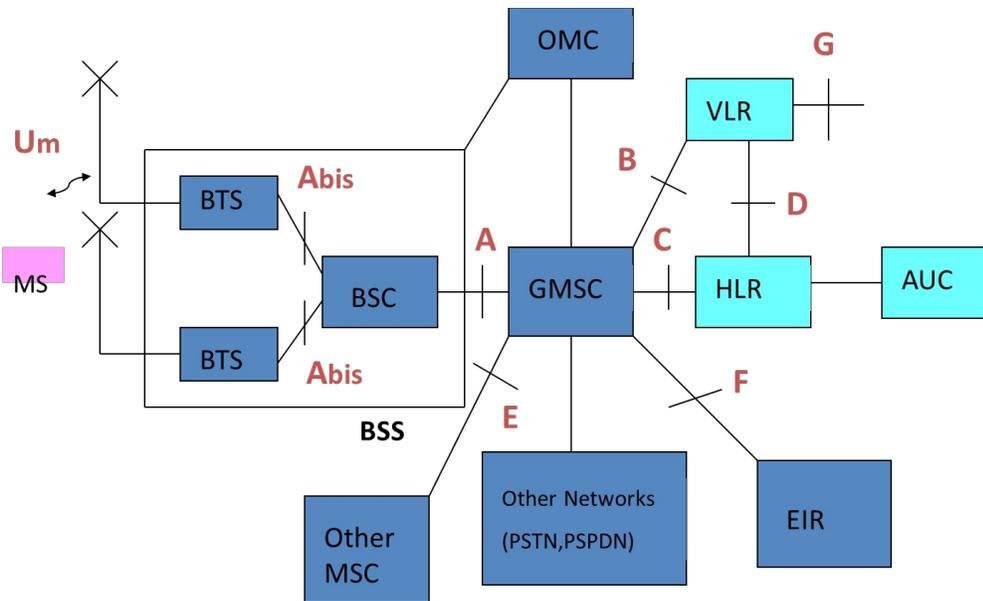


Fig 4: GSM Network Architecture

**I). Base station subsystem**

The base station subsystem (BSS) is the section of a traditional cellular telephone network which is responsible for handling traffic and signaling between a mobile phone and the network switching subsystem. The BSS carries out transcoding of speech channels, allocation of radio channels to mobile phones, paging, transmission and reception over the air interface and many other tasks related to the radio network.

## **Base transceiver station**

The base transceiver station, or BTS, contains the equipment for transmitting and receiving radio signals (transceivers), antennas, and equipment for encrypting and decrypting communications with the base station controller (BSC). Typically a BTS for anything other than a picocell will have several transceivers (TRXs) which allow it to serve several different frequencies and different sectors of the cell (in the case of sectorised base stations).

A BTS is controlled by a parent BSC via the "base station control function" (BCF). The BCF is implemented as a discrete unit or even incorporated in a TRX in compact base stations. The BCF provides an operations and maintenance (O&M) connection to the network management system (NMS), and manages operational states of each TRX, as well as software handling and alarm collection. The functions of a BTS vary depending on the cellular technology used and the cellular telephone provider. There are vendors in which the BTS is a plain transceiver which receives information from the MS (mobile station) through the Um (air interface) and then converts it to a TDM (PCM) based interface, the Abis interface, and sends it towards the BSC. There are vendors which build their BTSs so the information is preprocessed, target cell lists are generated and even intracell handover (HO) can be fully handled. The advantage in this case is less load on the expensive Abis interface.

The BTSs are equipped with radios that are able to modulate layer 1 of interface Um; for GSM 2G+ the modulation type is GMSK, while for EDGE-enabled networks it is GMSK and 8-PSK. Antenna combiners are implemented to use the same antenna for several TRXs (carriers), the more TRXs are combined the greater the combiner loss will be. Up to 8:1 combiners are found in micro and pico cells only. Frequency hopping is often used to increase overall BTS performance; this involves the rapid switching of voice traffic between TRXs in a sector. A hopping sequence is followed by the TRXs and handsets using the sector. Several hopping sequences are available, and the sequence in use for a particular cell is continually broadcast by that cell so that it is known to the handsets.

A TRX transmits and receives according to the GSM standards, which specify eight TDMA timeslots per radio frequency. A TRX may lose some of this capacity as some information is required to be broadcast to handsets in the area that the BTS serves. This information allows the handsets to identify the network and gain access to it. This signalling makes use of a channel known as the Broadcast Control Channel (BCCH).

## **Alarms in BTS**

There are two classes of alarms in BTS:

**External Alarms:** These alarms are external in nature and are caused due to environmental conditions or infrastructural failure

**Internal Alarms:** They are internal to the BTS system

### **External Alarm**

- **Power plant & Battery:-** Generally BTS are inside over exchanges so they take power from existing power plant if not at least 25-50 A module power plant with 200 AH battery set must be provided. The health of the battery is very crucial as at most of the site it is seen that BTS is off due to no battery backup and DG.
- **Engine Alternator:** Ensure the working of engine alternator make sure that battery of DG set is working properly and ensure the starting of DG when mains fails. Timely check the DG (periodically test).

### **Internal Alarm**

- BTS fail.
- TRX card faulty
- Coupler/Combiner faulty
- High coupler/combiner loss
- Swapping of feeder cable with adjacent sector.
- BCCH TRX failure
- High BER in PCM
- Power failure
- Media failure
- Fan failure Alarm

### **Base station controller**

The base station controller (BSC) provides, classically, the intelligence behind the BTSs. Typically a BSC has tens or even hundreds of BTSs under its control. The BSC handles allocation of radio channels, receives measurements from the mobile phones, and controls handovers from BTS to BTS (except in the case of an inter-BSC handover in which case control is in part the responsibility of the anchor MSC). A key function of the BSC is to act as a concentrator where many different low capacity connections to BTSs (with relatively low utilisation) become reduced to a smaller number of connections

towards the mobile switching center (MSC) (with a high level of utilisation). Overall, this means that networks are often structured to have many BSCs distributed into regions near their BTSs which are then connected to large centralised MSC sites.

The BSC is undoubtedly the most robust element in the BSS as it is not only a BTS controller but, for some vendors, a full switching center, as well as an SS7 node with connections to the MSC and serving GPRS support node (SGSN) (when using GPRS). It also provides all the required data to the operation support subsystem (OSS) as well as to the performance measuring centers.

A BSC is often based on a distributed computing architecture, with redundancy applied to critical functional units to ensure availability in the event of fault conditions. Redundancy often extends beyond the BSC equipment itself and is commonly used in the power supplies and in the transmission equipment providing the A-ter interface to PCU.

The databases for all the sites, including information such as carrier frequencies, frequency hopping lists, power reduction levels, receiving levels for cell border calculation, are stored in the BSC. This data is obtained directly from radio planning engineering which involves modelling of the signal propagation as well as traffic projections.

### **Transcoder**

The transcoder is responsible for transcoding the voice channel coding between the coding used in the mobile network, and the coding used by the world's terrestrial circuit-switched network, the Public Switched Telephone Network. Specifically, GSM uses a regular pulse excited-long term prediction (RPE-LTP) coder for voice data between the mobile device and the BSS, but pulse code modulation (A-law or  $\mu$ -law standardized in ITU G.711) upstream of the BSS. RPE-LPC coding results in a data rate for voice of 13 kbit/s where standard PCM coding results in 64 kbit/s. Because of this change in data rate for the same voice call, the transcoder also has a buffering function so that PCM 8-bit words can be recoded to construct GSM 20 ms traffic blocks.

Although transcoding (compressing/decompressing) functionality is defined as a base station function by the relevant standards, there are several vendors which have implemented the solution outside of the BSC. Some vendors have implemented it in a stand-alone rack using a proprietary interface. In Siemens' and Nokia's architecture, the transcoder is an identifiable separate sub-system which will normally be co-located with the MSC. In some of Ericsson's systems it is integrated to the MSC rather than the BSC. The reason for these designs is that if the compression of voice channels is done at the site of the MSC, the number of fixed transmission links between the BSS and MSC can

be reduced, decreasing network infrastructure costs. This subsystem is also referred to as the transcoder and rate adaptation unit (TRAU). Some networks use 32 kbit/s ADPCM on the terrestrial side of the network instead of 64 kbit/s PCM and the TRAU converts accordingly. When the traffic is not voice but data such as fax or email, the TRAU enables its rate adaptation unit function to give compatibility between the BSS and MSC data rates.

## **II). Network switching subsystem**

Network switching subsystem (NSS) (or GSM core network) is the component of a GSM system that carries out call switching and mobility management functions for mobile phones roaming on the network of base stations. It is owned and deployed by mobile phone operators and allows mobile devices to communicate with each other and telephones in the wider Public Switched Telephone Network or (PSTN). The architecture contains specific features and functions which are needed because the phones are not fixed in one location.

The NSS originally consisted of the circuit-switched core network, used for traditional GSM services such as voice calls, SMS, and circuit switched data calls. It was extended with an overlay architecture to provide packet-switched data services known as the GPRS core network. This allows mobile phones to have access to services such as WAP, MMS, and the Internet.

All mobile phones manufactured today have both circuit and packet based services, so most operators have a GPRS network in addition to the standard GSM core network.

### **Mobile switching center (MSC)**

The mobile switching center (MSC) is the primary service delivery node for GSM/CDMA, responsible for routing voice calls and SMS as well as other services (such as conference calls, FAX and circuit switched data). The MSC sets up and releases the end-to-end connection, handles mobility and hand-over requirements during the call and takes care of charging and real time pre-paid account monitoring. In the GSM mobile phone system, in contrast with earlier analogue services, fax and data information is sent directly digitally encoded to the MSC. Only at the MSC is this re-coded into an "analogue" signal (although actually this will almost certainly mean sound encoded digitally as PCM signal in a 64-kbit/s timeslot, known as a DS0 in America). There are various different names for MSCs in different contexts which reflects their complex role in the network, all of these terms though could refer to the same MSC, but doing different things at different times.

The Gateway MSC (G-MSC) is the MSC that determines which visited MSC the subscriber who is being called is currently located at. It also interfaces with the PSTN.

All mobile to mobile calls and PSTN to mobile calls are routed through a G-MSC. The term is only valid in the context of one call since any MSC may provide both the gateway function and the Visited MSC function, however, some manufacturers design dedicated high capacity MSCs which do not have any BSSs connected to them. These MSCs will then be the Gateway MSC for many of the calls they handle.

The visited MSC (V-MSC) is the MSC where a customer is currently located. The VLR associated with this MSC will have the subscriber's data in it. The anchor MSC is the MSC from which a handover has been initiated.

### **Mobile switching centre server (MSCS)**

The mobile switching centre server is a soft-switch variant of the mobile switching centre, which provides circuit-switched calling, mobility management, and GSM services to the mobile phones roaming within the area that it serves. MSS functionality enables split between control (signalling) and user plane (bearer in network element called as media gateway/MG), which guarantees better placement of network elements within the network. MSS and MGW media gateway makes it possible to cross-connect circuit switched calls switched by using IP, ATM AAL2 as well as TDM. More information is available in 3GPP TS 2205.

### **The MSC connects to the following elements:**

- The home location register (HLR) for obtaining data about the SIM and mobile services ISDN number (MSISDN; i.e., the telephone number).
- The base station subsystem which handles the radio communication with 2G and 2.5G mobile phones.
- The UMTS terrestrial radio access network (UTRAN) which handles the radio communication with 3G mobile phones.
- The visitor location register (VLR) for determining where other mobile subscribers are located.
- Other MSCs for procedures such as handover.

## **Broadband**

The term broadband refers to a telecommunications signal or device of greater [bandwidth](#), in some sense, than another standard or usual signal or device (and the broader the band, the greater the capacity for traffic).

Prior to the invention of home broadband, dial-up internet was the only means by which one could download songs, movies, e-mails, etc. Unfortunately, it would take up to 10-30 minutes to download one song (5MB) and over 28 hours to download a movie (700MB). Dial-up internet was also extremely inconvenient since it took up the use of the home telephone line, and homes would have to decide if paying for a second telephone line was worth its cost.

The cable modem was the first broadband option available, but due to the small amount of cable Internet subscribers for the first year in 1997, broadband didn't take off until 2001. Having home broadband made downloading times 10X faster than dial-up. Unfortunately, like many new technologies, most consumers were unable to afford such a luxury of fast internet. Price barriers weren't a factor for long, and by 2004 the average American households considered home broadband to be affordable. Since its creation, broadband has continually strengthened and available speeds have become faster and faster.

Different criteria for "broad" have been applied in different contexts and at different times. Its origin is in physics, [acoustics](#) and radio systems engineering, where it had been used with a meaning similar to wideband. However, the term became popularized through the 1990s as a vague marketing term for [Internet access](#).

## **INTRODUCTION TO NATIONAL INTERNET BACKBONE-II (NIB-II)**

### **1.SCOPE**

This Engineering Instruction presents an introduction to National Internet Backbone-II (NIB-II) conceived by BSNL to provide infrastructure for providing number of value added services to broadband customers countywide with guaranteed quality of service (QoS).

### **2.GENERAL**

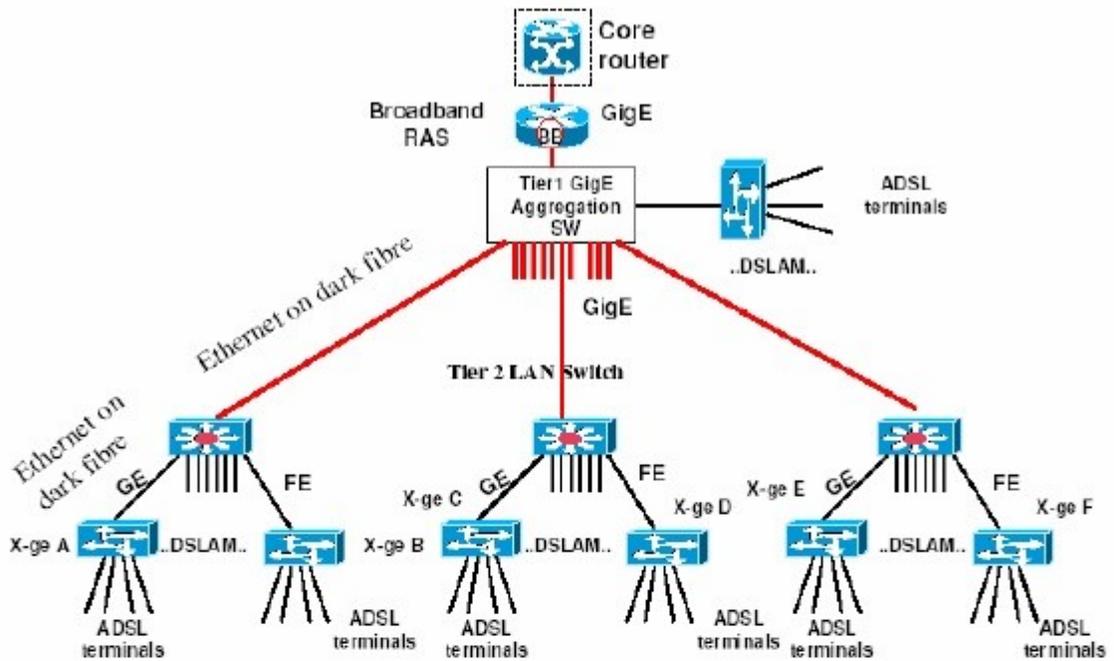
BSNL has planned to setup NIB-II to provide world class infrastructure to offer various value added services to a broader customer base county-wide that will help to accelerate the Internet revolution in India. Moreover the NIB-II will create a platform, which enables e-governance, e-banking, e-learning, etc. with the key point of Service Level Agreements & Guarantee in tune with Global standards and customer expectations. NIB-II has been grouped into following three major projects.

Fig. 5: Core Network Architecture for NIB-II  
 Fig 6: Broadband Connectivity in City

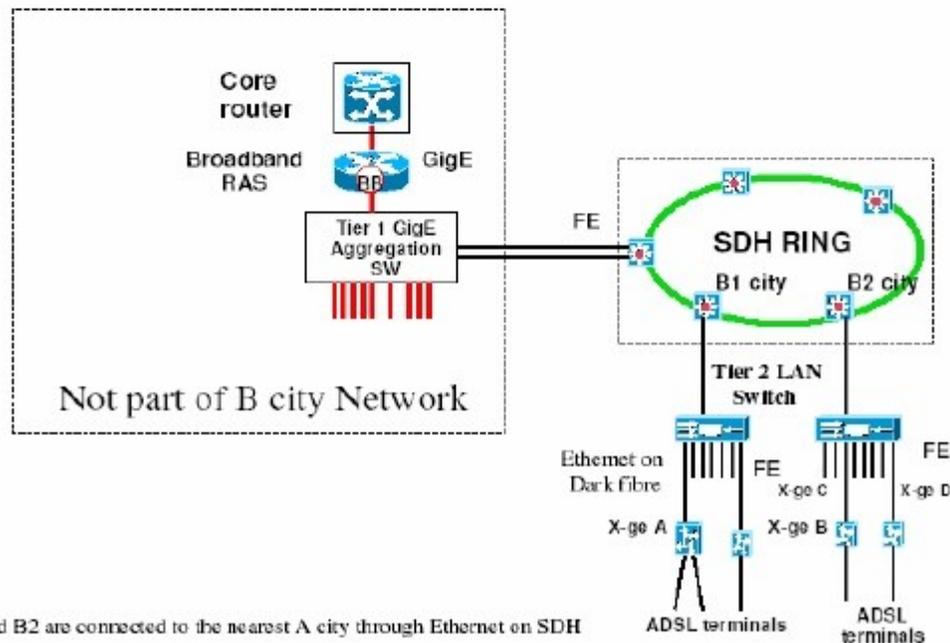
Modem

A modem (modulator-demodulator) is a device that modulates an [analog carrier signal](#) to encode [digital information](#), and also demodulates such a carrier signal to decode the transmitted information. The goal is to produce a [signal](#) that can be transmitted easily

## Broadband Connectivity in A city



## Broadband Connectivity in B City



Note: B1 and B2 are connected to the nearest A city through Ethernet on SDH

and decoded to reproduce the original digital data. Modems can be used over any means of transmitting analog signals, from [light emitting diodes](#) to [radio](#). The most familiar example is a [voice band](#) modem that turns the [digital data](#) of a [personal computer](#) into modulated [electrical signals](#) in the [voice frequency](#) range of a [telephone channel](#). These signals can be transmitted over [telephone lines](#) and demodulated by another modem at the receiver side to recover the digital data.

Modems are generally classified by the amount of data they can send in a given [unit of time](#), usually expressed in [bits per second](#) (bit/s, or bps). Modems can alternatively be classified by their [symbol rate](#), measured in [baud](#). The baud unit denotes symbols per second, or the number of times per second the modem sends a new signal. For example, the ITU V.21 standard used audio frequency-shift keying, that is to say, tones of different frequencies, with two possible frequencies corresponding to two distinct symbols (or one bit per symbol), to carry 300 bits per second using 300 baud. By contrast, the original ITU V.22 standard, which was able to transmit and receive four distinct symbols (two bits per symbol), handled 1,200 bit/s by sending 600 symbols per second (600 baud) using [phase shift keying](#).

#### DSLAM

A digital subscriber line access multiplexer (DSLAM, often pronounced dee-slam) is a network device, often located in the [telephone exchanges](#) of the telecommunications operators. It connects multiple customer [digital subscriber line](#) (DSL) interfaces to a high-speed digital communications channel using [multiplexing](#) techniques.<sup>[1]</sup>

By placing additional DSLAMs at locations remote from the [telephone exchange](#), [telephone companies](#) provide DSL service to locations previously beyond effective range.

#### Broadband Network Gateway(BNG)

A network gateway is an internetworking system capable of joining together two networks that use different base protocols. A network gateway can be implemented completely in software, completely in hardware, or as a combination of both. Depending on the types of protocols they support, network gateways can operate at any level of the [OSI model](#).

## CONCLUSION

The final conclusion is the study of basics of TELECOMMUNICATION system installed and operated in BSNL will help us understand the practical variations in the theoretical concepts studied in our course plan. The study also helps us to have a better understanding of the current technology being implemented and also the scope of further research in the field of communication. As we are moving more towards the software and IT sector, even communication sector is evolving itself into a software based hardware sector. So the further scope for research and development is vast.

## **REFERENCES**

1. <https://www.google.com>
2. <https://en.wikipedia.org>
3. [www.radio-electronics.com](http://www.radio-electronics.com)
4. Hand written notes