QUESTION 16/2

Preparation of handbooks for developing countries

ITU-D STUDY GROUP 2 2nd STUDY PERIOD (1998-2002)

Handbook on new technologies and new services

FASCICLE 1
New technologies supporting new networks

Telecommunication Development Bureau (BDT)

International Telecommunication Union

THE STUDY GROUPS OF THE ITU-D

The ITU-D Study Groups were set up in accordance with Resolution 2 of World Telecommunication Development Conference (WTDC) held in Buenos Aires, Argentina, in 1994. For the period 1998-2002, Study Group 1 is entrusted with the study of eleven Questions in the field of telecommunication development strategies and policies. Study Group 2 is entrusted with the study of seven Questions in the field of development and management of telecommunication services and networks. For this period, in order to respond as quickly as possible to the concerns of developing countries, instead of being approved during the WTDC, the output of each Question is published as and when it is ready.

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FASCICLE 1
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Telecommunication Development Bureau (BDT)
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FASCICLE 1

New technologies supporting new networks

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Note for the attention of the reader

This handbook has been prepared by many volunteers from different Administrations and companies. They provided examples of their products, systems, models and case studies.

The mention of specific companies or products does not imply any endorsement or recommendation by ITU.

PREFACE

1 Historical background

In today's Global Information Society (GIS) and Global Information Economy (GIE), the telecommunication sector is expected to play a major role as a leading industry in the 21st century. To meet the requirements of this new century and to close the communication gap between the industrialized and developing countries, the sharing of knowledge in telecommunication technology and services is a very important step. This was recognized by the first World Development Telecommunication Conference held in Buenos Aires, 21-29 March 1994, and also confirmed by the second World Telecommunication Development Conference held in Valletta, 23 March-1 April 1998, which established two Study Groups:

- Study Group 1: dealing with telecommunication development strategy and policies.
- Study Group 2: dealing with development and management of telecommunication services and networks.

The particular Question 2/2 in the first Study Period and Question 16/2 in the present Study Period of Study Group 2 has the objective to prepare new handbooks or revise the existing ones to spread knowledge and know-how in these fields. The handbook "New Technologies and New Services" is one way to achieve this objective.

2 Purpose of the handbook

The rapid development of telecommunications, from technological to market aspects, brings with it every day new products, equipment, systems, networks and services. It would be somewhat ambitious to attempt to produce a handbook that comprises all aspects of new technologies and new services in telecommunication matters and that fulfils, the needs of all players in the telecommunication arena.

The aim of this handbook is to provide a survey of technologies and new services taking place in the changing telecommunication environment, by presenting the general characteristics and capabilities that various networks and new services offer on the market, although not dealing with technical details that are the subject of standardization. The reviewed technologies and new services are in general compliant with the ITU Recommendations.

The handbook also touches on gender perspectives in the preparation and introduction of new services.

3 Why is the handbook necessary?

The telecommunication sector has gone through radical changes in the last decade, driven by an increasingly global and liberalized market in which the control of network capabilities has become a strategic competitive factor to satisfy increased customer requirements. The fast evolution of network intelligence is mainly driven by the convergence between telecommunications, computers and information technology developing various multimedia services. Due to that, telecommunication networks are becoming and will become increasingly complex and challenging to implement and operate. Moreover, it will be more and more important to possess the competence necessary to integrate high capacity and intelligent solutions into the existing networks to meet end-user and cost efficiency requirements.

Three of the most important requirements that have to be met by the improvement of the existing or newly planned networks are:

- *more capacity*,
- *more power,* and
- *greater efficiency.*

To summarize these driving forces, and also to provide a basis for the subject of the handbook, we can say that:

- With insufficient capacity (e.g. bandwidth, packet volume, etc.), network operators cannot satisfy the users demands, even for the universal/service access and especially for new and high quality services. However, in order to optimize the investment, more creativity and appropriate planning of the new technology applications are required.
- The opportunity of new technology applications to build up more intelligent capabilities in telecommunication equipment will give more power to network operators and service providers while increasing Intelligent Network traffic, generating new value-added services. Estimation of customers' needs is a powerful tool to win competition, but only if it is done in time.
- Development of new network management concepts, comprising operation, administration, maintenance and planning, leads to better operational efficiency and maintenance organization, thereby decreasing operational costs. An objective picture of network elements contained in, networks, services and business will improve day-to-day operations and network performance, thus fulfilling customers' expectations for higher quality of service as the target.

These reasons have been the leading concepts in developing the material contained in this handbook.

4 Who should read the handbook?

The handbook is a convenient tool for all those who are interested in telecommunications. It is specially recommended to managers, planning and technical experts of incumbent telecommunication operators as well as national regulators in the telecommunication sector, in particular from developing countries. Those managers should use this information to overview or develop concepts for long-term strategies. Technical managers and experts should extract the deployment of networks that will support the new and global services expected by the 21st century users. Regulators, especially newly established ones, should be supported to create an environment in which innovations by various players can be encouraged without applying constraints other than those needed to create productive competition. Other players will also benefit from the prepared material for improvement of their market access.

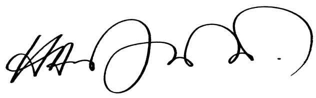
References listed in the various fascicles of the handbook should enable readers to obtain more information on the subject.

5 Homogeneity

Many people have contributed to this handbook which was edited by Ms N. Gospic assisted by Messrs B. Moore and J. Magill. The chapters and subchapters vary in scope and level of detail which is to be expected in such a handbook, reflecting the different nature of the contributions.

The handbook represents a snapshot of the development of networks and services as at the time of issue. It is therefore not a complete picture but provides guidelines and many references for further study.

It is anticipated that the handbook will be updated as the technologies evolve.



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EXECUTIVE SUMMARY

1 Introduction

Having regard to future technical and operational trends in telecommunication services, the developing countries need to know about the latest advances made by the international community in this regard, in order to pass on the benefits to their own nationals.

The developing countries must keep abreast of developments and achievements in the application of new technologies in telecommunication networks, in order to pass on the benefits of this progress to their users.

The handbook on new technologies and new services has been prepared taking into account these two statements of the Valletta Conference held in 1998.

2 Handbook structure

The complexity of telecommunication matters, and the different groups of readers, require the handbook structure to follow a certain pedagogical pattern. It is intended that the reader will be able to easily find information on a certain part of a subject. The structure is also adapted to the fast changes in telecommunications and to the fact that some subjects are still under study.

The handbook consists of four parts, which are prepared as separate fascicles:

Fascicle 1: "New technologies supporting new networks"

Fascicle 2: "Digital networks and services"

Fascicle 3: "IP-based networks and services"

Fascicle 4: "Digital radio and television networks and services"

Each chapter is self-contained (and in some cases subchapters are self-contained) in order to simplify updating of the text.

The mention of specific companies in this handbook does not imply any endorsement or recommendation by ITU.

In each fascicle there is a Chapter 1 dealing with a general review and links to other fascicles. Relevant ITU-T standards and publications are referenced in each of the chapters. Also, some of the important standards from other standardization organizations are listed. In some chapters, useful examples are annexed in order to amplify the subject matter.

A brief content of the fascicles as part of the handbook is given below. The detailed table of contents is shown at the beginning of each fascicle.

Fascicle 1 – New technologies supporting new networks

The content of this fascicle is divided into the following chapters:

1 Introduction

2 New transport media technologies
   - Optical fibre cables
   - Digital radio-relay technologies
   - Mobile communication systems
   - Satellite systems

3 Digital switching systems
4 New signalling systems and SS No. 7
5 Synchronization techniques and methods
6 Digital transmission
7 ATM Technology

Chapter 1 – Introduction and general review, discusses the need to implement new technologies in order to introduce new services and be competitive on the market with capacities and quality.

Chapter 2 – New transport media technologies, deals with main considerations when deploying optical, digital, radio, and satellite technologies.

Chapter 3 – Digital switching systems, deals with technologies for circuit and packet switching and SPC switching system organization.

Chapter 4 – New signalling systems and SS No. 7, includes necessary specifications for new digital networks.

Chapter 5 – Synchronization techniques and methods, explains implementation of synchronization in new digital networks.

Chapter 6 – Digital transmission systems, deals with PDH, SDH, WDM and xDSL techniques with reference to the most important standards and implementation examples.

Chapter 7 – ATM technology, includes ATM transport, switching and cell format, operation and maintenance and signalling and traffic management of ATM networks.

Fascicle 2 – Digital networks and services

Fascicle 2 is composed of eight chapters, annexes and test cases:

1 Introduction
2 Digital networks and services
3 Mobile digital cellular networks and services
4 Access networks
5 Network and service management
6 Planning aspects
7 Human resources aspects
8 Financial and economic aspects

Chapters 2, 3 and *4* – deal with different network technologies and structure and related services, emphasizing their main characteristics and requirements for new architectures and interworking. Chapter 2 is divided in 10 subchapters dealing with particular fixed networks, e.g. PSTN, ISDN, IN, Packet switched network, Frame relay, ATM-based networks, services and ITU-T Standards for related subjects. Development of China Telecommunications is included as Annex 2A.

Chapter 5 – deals with Service and Network Management based on the introduction of the TMN concept.

Chapter 6 – offers the guidelines for network planning with annexes as examples.

Chapter 7 – is concerned with development of human resources being capable to implement new technologies and new services, and

Chapter 8 – deals with financial and economic aspects in developing the new networks and services.

Fascicle 3 – IP-based networks and services

Fascicle 3 is composed as follows:

1 Table of contents

2 Introduction and definitions

3 Internet Protocol (IP)

4 E-commerce

5 TeleInternet services for Tel-E-Commerce

Chapter 2 – is based on dramatic grow of Internet subscribers, new services and IP-based networks. It deals with basic definitions for E-mail, WWW, Arpanet and hyperlinks.

Chapter 3 – deals with Internet Protocol-IP characteristics, IP packet structure, IP address, Voice over IP and Ipv4 and Ipv6.

Chapter 4 – explains E-commerce as a new data service realizing the vision of the Global Information Economy.

Chapter 5 – deals with new IP architecture and applications of TeleInternet services with reference to ITU publications.

Fascicle 3 is an initial introduction to the subject and further revision and expansion will be required.

Fascicle 4 – Digital radio and television networks and services

This fascicle contains five chapters:

1 Introduction

2 Digital audio broadcasting

3 Digital television broadcast services

4 Strategies for digital television broadcasting

5 Data broadcasting

Chapter 1 – deals with general introduction about radio and television networks and services.

Chapter 2 – summarizes benefit of audio digital broadcasting introducing different services and digital audio systems. It deals mainly with the terrestrial digital audio broadcasting system (T-DSB).

Chapter 3 – lists advantages of digital TV transmission, both Standard Definition Television and High Definition Television. The chapter discusses the structure of digital television system, digital satellite broadcasting, digital terrestrial broadcasting, planning aspects, different standards and networks and services.

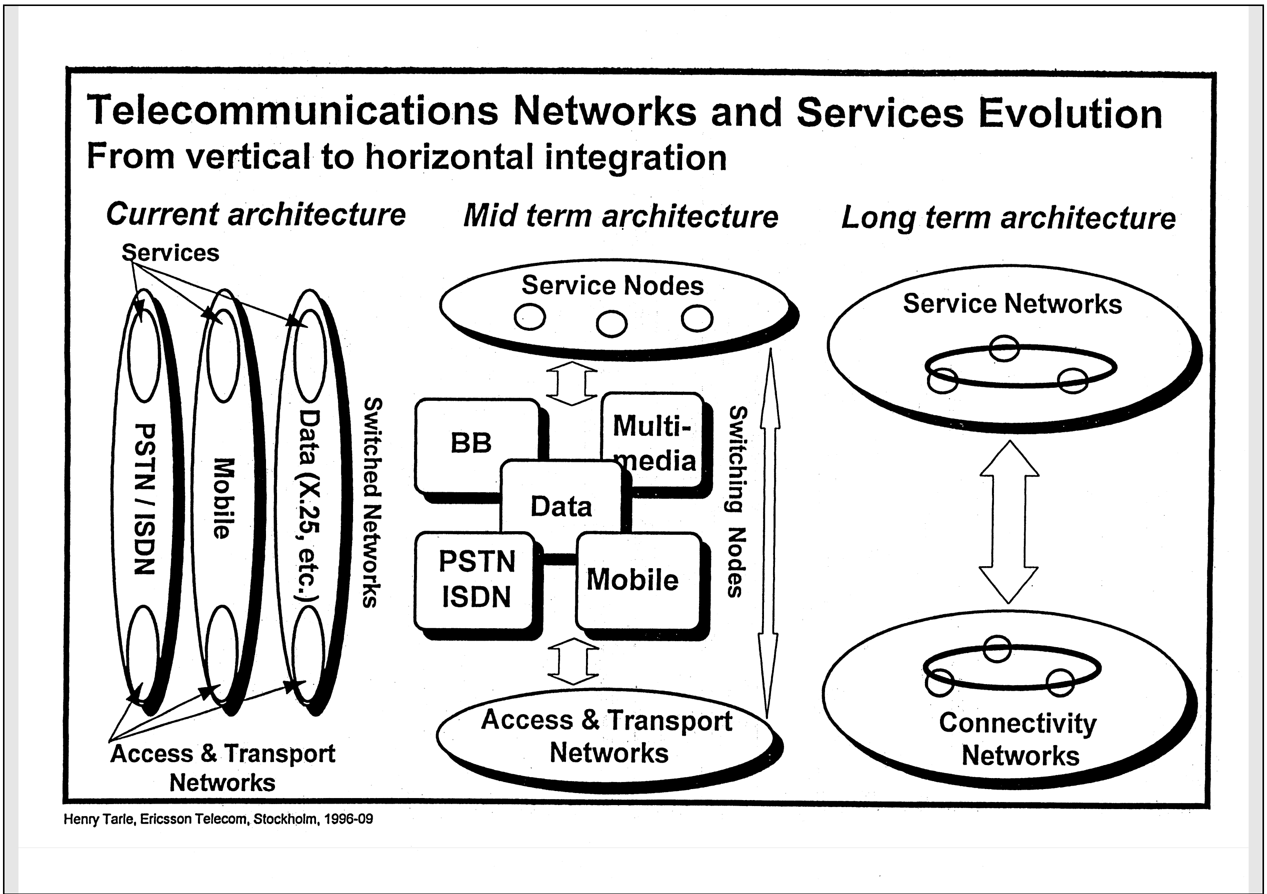
Chapter 4 – deals with strategies for digital television broadcasting emphasizing the needs in transition from analogue to digital systems. The chapter identifies planning criteria for different systems, networks and services.

Chapter 5 – "Data broadcasting" exploits a new development area for broadcasters in a competitive environment. Data broadcasting services are defined with requirements for data systems. The chapter deals in more detail with data broadcasting over terrestrial systems using broadband wireless, multichannel multipoint distribution system, ISDN, cable TV distribution and some examples for multimedia broadcasting.

How to make use of the handbook

To make a good use of the handbook, it is necessary to understand the changes in the managing of the telecommunication business. Three categories are very important for successful management of telecommunication:

Figure – Telecommunication networks and services evolution (From Vertical to Horizontal Integration)



How to read the handbook

The following charts are aimed at facilitating the reading of the handbook for different groups of readers:

Figure – For telecom operators and regulator managers

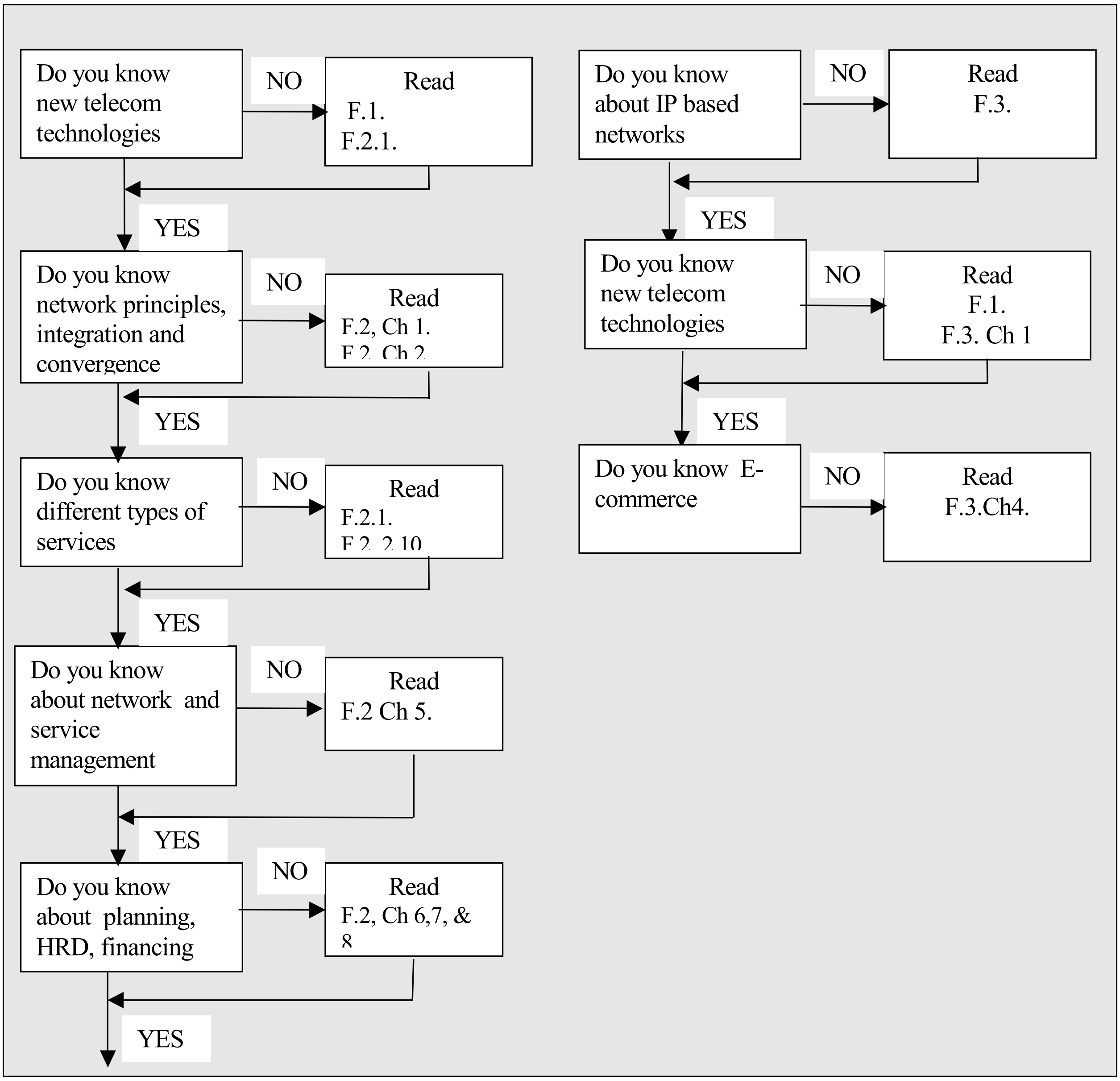


Figure – For broadcast managers and operator staff

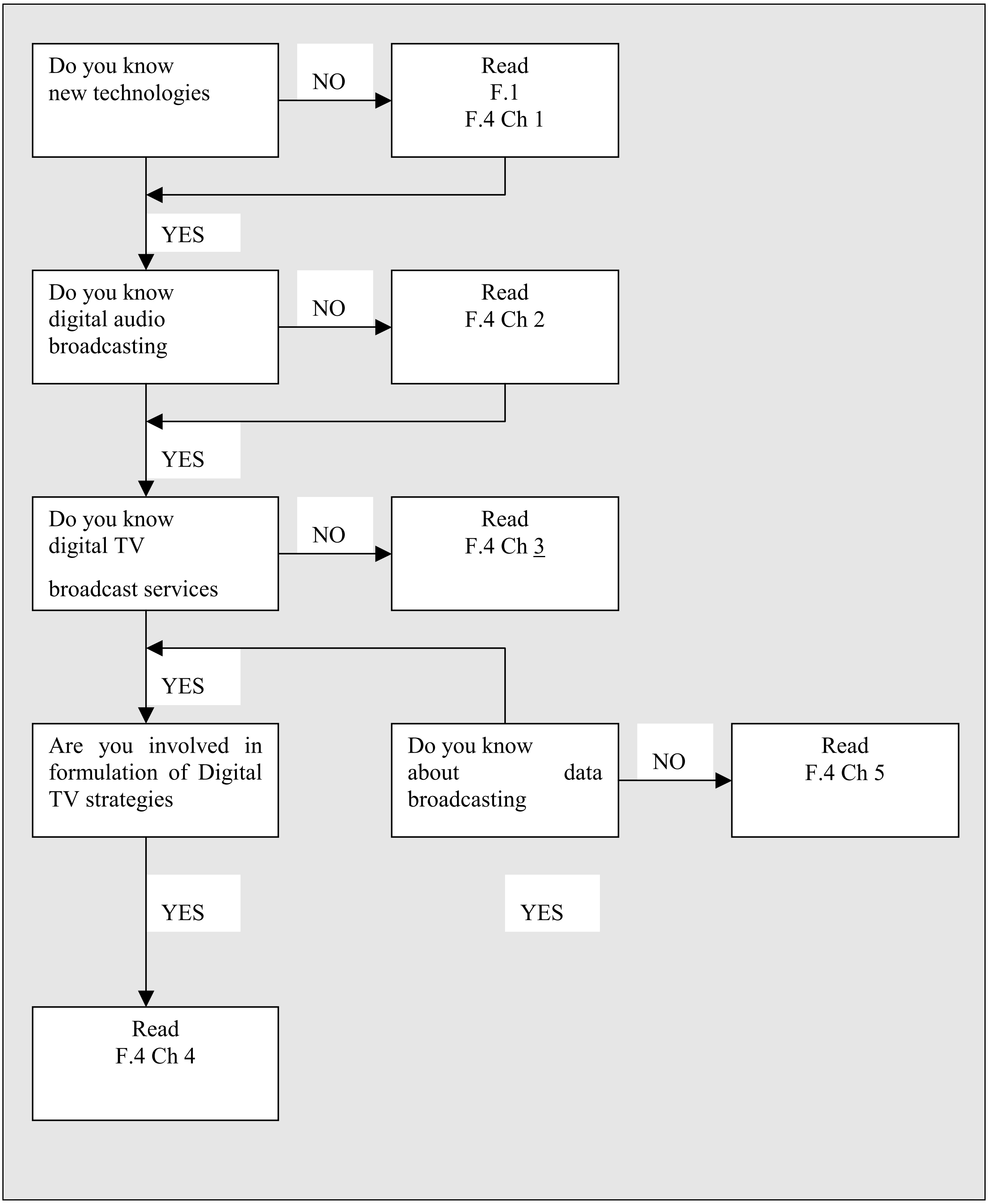
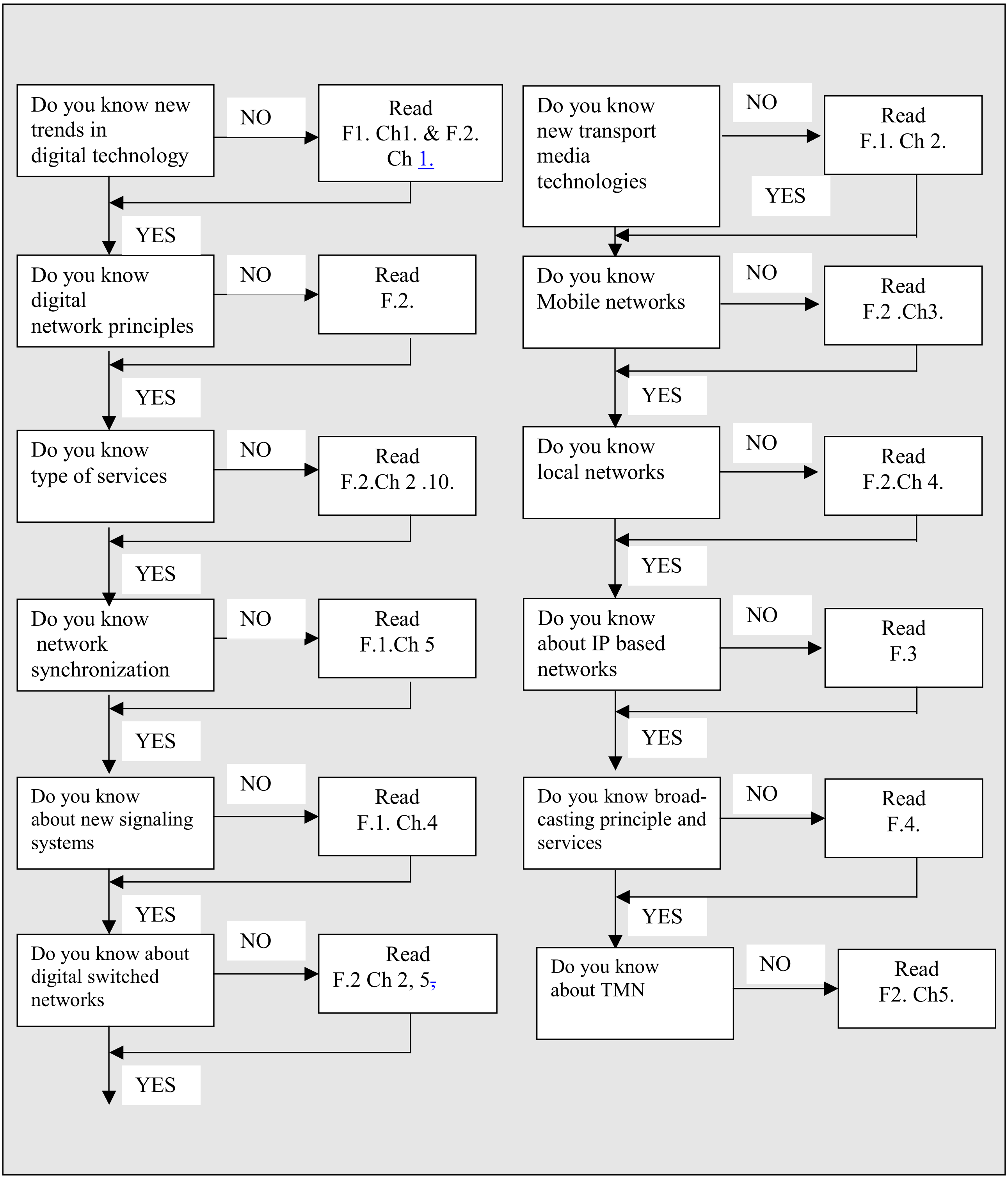


Figure – For planners and technical managers



CHAPTER 1

Introduction

1.1 General

Extensive demand for voice and data communication is offering a major opportunity for telecommunication operators. Demand for broadband services, Internet access with the need for high speed data transmission, Video on Demand etc. require significantly increasing capacities in the network, challenging operators of fixed and mobile networks to deploy enhancements to their infrastructures quickly enough to respond for those demands. In addition, operators of narrowband wireless local loop and cellular networks require cost-effective transmission feeders to connect their base stations. A powerful infrastructure with appropriate transmission media will enable operators to be prepared for emerging new trends in networks and services. In many cases, the telecommunication infrastructure consists of a mix of different technologies for the successful realization of physical and logical networks.

Optical fibre is recognized as the superior transmission media and consequently has already been in use for a number of years. It supports the development of a new digital hierarchy e.g. the Synchronous Digital Hierarchy, SDH. The global demand for optical fibre cables is expected to grow at between 15 to 20% per year over the coming years. Thirty six million fibre/kilometres were installed in 1997. Optical fibre deployment started in long distance networks, but the market for optical cables in the local telephone, cable TV and data communication applications continues to grow, more and more rapidly. Optical fibre cables are installed by either burial (directly in underground ducts) or on poles. The new developments in the field of the Wavelength Division Multiplexing – WDM, have emphasized the role of using optical fibres over long time periods, and demonstrate the possibility of further exploiting the existing fibre optical networks. Thanks to the new techniques, such as optical amplifiers and Dense Wavelength Division Multiplexing – DWDM, the cost of the network infrastructure is significantly cheaper than in the past.

New optoelectronic devices, studied in this Fascicle, will make it possible to build optical networks with high traffic capacities operating at high bit rates (several hundred Gbit/s to Terabit/s).

However, great progress has also been made in the development of radio technologies, specially, in satellite and cellular technologies, changing dramatically the fields of telecommunication development, universal access and user mobility. In order to facilitate implementation of these different telecommunication technologies the ITU has issued different publications elaborating these matters. This Fascicle, which is a part of the Handbook "New Technologies and New Services" has such a role.

1.2 Structure of Fascicle 1

Fascicle 1 is divided in 7 Chapters with this Introduction as Chapter 1.

Each Chapter is self-contained (apart from Chapter 2 where each sub-chapter is self-contained) in order to simplify the updating of the text.

Chapter 2 – *New transport media technologies,* is divided in four Subchapters: Optical Fibres, Radio-Relay Systems, Mobile Communication Systems, and Satellite Systems.

The first Subchapter, *Optical Fibres*, deals with the main considerations when deploying optical fibres, discussing transmission properties (e.g. Attenuation, Chromatic dispersion, and polarization), fibre types, optical fibre implementation, and protection of optical fibre rings.

Digital Radio-relay Systems, as Subchapter 2, deals with developments in digital microwave technologies, multiple access radio systems, and digital radio networks.

The third Subchapter addresses *Mobile Communications Systems* including spectrum allocation, existing analogue cellular systems, and the various digital cellular systems. Also discussed are cordless telephony, and the use of satellite systems. The subject of third generation mobile systems is primarily addressed in Fascicle 2.

The fourth and final Subchapter, *Satellite Systems*, discusses the satellite technologies as a special case of radio technology. Satellite systems are similar to radio systems, the only real difference being that the intermediate link station is in orbit. Satellites used for telecommunications purposes are placed in geostationary, medium earth and low earth orbits. The development of global satellite based personal communication systems – GMPCS, is addressing a new market, which has not been exploited by costly conventional mobile satellite systems. The new satellite techniques, such as VSAT, DAMA, TDMA, DCME, are discussed also. As examples, Annexes introduce global satellite systems such as Globstar, ICO, SkyBridge, Teledesic and Final Analysis.

Chapter 3 – *Digital Switching Systems*, deals with technologies for circuit and packet switching, and with SPC switching system organization. The development of SPC technology has made it possible to design nodes with extensive functionality to carry different services. These capabilities are called “network intelligence” and are the subject of this subchapter. New switching procedures, such as ATM and optical switching, are discussed here also.

Chapter 4 – studies *New signalling Systems and SS No. 7* and is composed of four Subchapters. The Introduction highlights the importance of new signalling systems for network evolution and new services.

Subchapter 2 deals with Signalling System No. 7 with details on the Message Transfer Part (MTP), Signalling Connection Control Part (SCCP), Transaction Capabilities (TC), ISDN User Part (ISUP), and the Intelligent Network Application Part (INAP).

Subchapter 3 deals with Digital Subscriber Signalling System 1 (DSS1).

Subchapter 4 evaluates Broadband signalling explaining User Network Interface (UNI), Broadband ISDN user Part (B-ISDN), Digital Signalling System Number 2 (DSS2) and relevant ITU-T Recommendations. Annexes deal with Signalling functions, Other ATM Signalling Protocols, and DSS2 and the ATM Forum UNI signalling specifications.

Chapter 5 – deals with *Synchronization Techniques and Methods* with Synchronization of the transmission systems, Phased Locked Loop, Timing correction, Network synchronization, and Synchronization in ATM are elaborated in separate Subchapters.

Chapter 6 – deals with transmission technology *PDH, SDH, DWDM and xDSL*. The Chapter introduces Synchronous Digital Hierarchy, SDH, explaining a SDH frame structure, multiplexing scheme, and SDH management. DWDM, as a new technology offers high transmission rates on fibres and capacity can be increased without laying new cables. In addition Chapter 6 also deals with the new transmission technology for upgrading cooper wire using – Digital Subscriber Line transmission techniques. A number of new technologies, classed as –‘xDSL’ technologies, offer new approaches to use the existing local networks for higher bit rate services.

Chapter 7 – deals with *ATM technology*. ATM frame structure, multiplexing, protocols and management are addressed in detail.

The references listed in the chapters should enable readers to get more information on the subjects.

The mention of specific companies or products in this Fascicle does not imply any endorsement or recommendation by the ITU.

CHAPTER 2

New transport media technologies (optical fibres, radio, satellite)

2.1 Optical fibres

2.1.1 Transmission properties

A typical optical fibre is made up of a central cylindrical part, the core, in which the light is normally travelling, surrounded by an optical cladding used to guide the light into the core, i.e. to maintain as much as possible the optical power inside the core. These two parts, the core and the cladding, are normally made of ultra pure doped silica. For mechanical protection, these two parts are covered by a concentric primary coating made of a plastic material, which is very important for insuring a stable and reliable behaviour. The primary coating is also, in general, marked, i.e. by a colour code for fibre identification.

Definitions of these parameters as well as other parameters listed below and the way to measure them can be found on the following publications:

ITU-T Recommendation G.650:	Definitions and test methods for the relevant parameters of single-mode fibres.	
ITU-T Recommendation G.651:	Characteristics of a 50/125 μm multimode graded index optical fibre cable.	
IEC Publication 61931:	Fibre optic terminology	
IEC Publication 60793-1:	Optical fibres – Part 1	*Generic specification*
IEC Publication 60794-1:	Optical cables – Part 1	*Generic specification*.

For single-mode fibres, which are the only ones used for telecommunication applications (multimode fibres are used only for Local Area Networks and short distance applications), the main transmission characteristics are the attenuation, the chromatic dispersion, the cut-off wavelength and the polarization mode dispersion.

Numerical values related to these parameters which can be found in International Standards or Recommendations are the result of a consensus between the different members at the time of preparation of the Standard. Therefore and considering also the evolution of the technology, it is possible for some parameters to find different values for the best products available on the market.

Several fibre parameters, in particular transmission parameters, can be affected by the cabling and/or installation. For designing a transmission system, it is essential to consider only fibre parameters of installed cables and therefore, the values given hereafter shall be understood as applicable to cabled fibres.

2.1.1.1 Attenuation

Attenuation is defined as:

1 A decrease of electromagnetic power between two points.

2 The quantitative expression of power decrease which may be expressed by the ratio of the values at two points of a quantity related to power in a well defined manner.

Attenuation is certainly one of the most important parameter for an optical fibre as it is the most common limitation of the maximum distance for an optical link. It is normally given as an attenuation coefficient, i.e. attenuation per unit length, and is normally measured around specific wavelengths, 1310 and 1550 nm for standard single-mode fibres. During the last ten to fifteen years, the value of the attenuation coefficient has been continuously reducing due to the evolution of the technology and is now very close of the theoretical minimum value.

Maximum attenuation coefficient values specified for cabled fibres are presently 0.5 dB/km (1310 nm) and 0.4 dB/km (1550 nm). Values of 0.38 to 0.40 dB/km at 1310 nm and around 0.25 dB/km at 1550 nm can be found on the market.

2.1.1.2 Chromatic Dispersion

Chromatic dispersion is:

The spreading of a light pulse per unit source spectrum width in an optical fibre caused by different group velocities of the different wavelengths composing the source spectrum.

NOTE – The chromatic dispersion may be due to the following contributions: material dispersion, waveguide dispersion, profile dispersion.

It may also limit the maximum distance for an optical link especially for high bit rates which are now used more and more. For the basic single-mode fibres, minimum attenuation is obtained in the 1550 nm window, and minimum chromatic dispersion happens in the 1310 nm window. The try to solve this problem, new types of fibres combining minimum attenuation and reduced chromatic dispersion have been proposed and are discussed in the following chapter.

Specified value for G.652/B1.1 fibres is presently: S_{0max} (ps/nm^2 · km) $\leq$ 0.093.

2.1.1.3 Cut-off Wavelength

Cut-off wavelength is defined as:

The wavelength at which the ratio between the total power, including launching higher order modes, and the fundamental mode power has decreased to less than a specified value, the modes being substantially uniformly excited.

NOTE – This specified value is presently chosen to be 0.1 dB.

NOTE – The cut-off wavelength depends upon the measurement conditions and in particular on the sample length, its curvature and cabling.

NOTE – The cut-off wavelength is generally different from the theoretical cut-off wavelength that can be computed from the refractive index profile of the fibre. The theoretical cut-off wavelength is a less useful parameter for determining fibre performances in a telecommunication network.

This parameter is the one that insures the single mode behaviour of the fibre. It can be understood from the note 2 that cut-off wavelength is affected by the cabling process and possibly by installation conditions. Therefore, it is necessary to know deployment conditions of a fibre to evaluate its cut-off wavelength, which must be smaller than the operating wavelength of the system. Indication on the way to choose cut-off wavelength is given in IEC 60793-2 by the following note:

In general, there is no unique relationship between the fibre cut-off wavelength λ and that of cabled fibre λ. The necessary value of ë$_c$ for any particular system application depends upon the fibre, cable and system design (including system operating bit-rate and operating wavelength), together with the expected length and curvatures of repair cables. However, a general relationship may be determined between λ and λ for particular fibre cable designs. The required value for λ is still under study and will be indicated in the product specification on optical cables (IEC 60794-2).

For some applications, some users have preferred to set limits for λ below the envisaged operating wavelength; typical values within the range 1100-1280 nm have been adopted in such cases. For other applications, some users have chosen to permit λ to be as high as 1350 nm, relying on the effects of cable making and installation to yield values below the operating wavelength range.

2.1.1.4 Polarisation Mode Dispersion

Polarization Mode Dispersion (PMD) is:

The distortion of the transmitted signal attributable to the different velocities of the polarization components of the same mode and their dependence on wavelength.

NOTE – In straight, unstressed circularly symmetric fibres the polarization dispersion is negligible.

PMD has been recognized recently as a potential limitation for very high bit rate long haul links. The statistical nature of PMD creates some difficulty for an accurate prediction of its impact on the global link. The consequences of PMD are particularly important for long distance applications as it may impair the performance of a long length and high bit rate fibre optic system (e.g. 10 Gbit/s transport over 100 km). When designing this type of system, it is therefore recommended to use very low PMD fibres to avoid any risk of impairing the global link performance.

Recommended PMD value for cabled fibres is presently ≤ 0.5 ps.km$^{-1/2}$. Values ≤ 0.2 ps.km$^{-1/2}$ or lower can be found on the market.

2.1.2 Fibre types

For long distance telecommunication applications, only single-mode fibres are used, as they exhibit the lowest attenuation and the highest transmission capability. However, there are different types of single-mode fibres, corresponding to different applications, which are described in the following ITU Recommendations:

G.652 Characteristics of a single-mode optical fibre cable

G.653 Characteristics of a dispersion-shifted single-mode optical fibre cable

G.654 Characteristics of a 1550 nm wavelength loss – minimized single-mode optical fibre cable

G.655 Characteristics of a non-zero dispersion shifted single-mode optical fibre

IEC specifies the fibres according to the following categories:

B1.1 Dispersion unshifted for operation at 1310 nm

B1.2 Loss minimized for operation at 1550 nm

B2 Dispersion shifted for operation at 1550 nm

B3 Dispersion flattened for operation at 1310 nm and 1550 nm

B4 Non zero dispersion for operation at 1550 nm

Characteristics of these different categories of fibres are specified in IEC publication 60793-2:

Optical fibres – Part 2: *Product specifications*.

The most common product, in terms of the installed base since the beginning of the single-mode technology, is fibre category B1.1, corresponding to Recommendation G.652. It is presently this fibre which represents the major part of the market due to its capability of providing, at the most attractive cost, good performance adaptable to most of the telecommunication applications.

For B1.1 single-mode fibres, minimum attenuation is obtained in the 1550 nm window, and minimum chromatic dispersion happens in the 1310 nm window. It is therefore not possible to achieve at the same time an optimum behaviour from the point of view of both loss and dispersion. Dispersion shifted fibres (DSF, category B2 according to Recommendation G.653) have been developed to answer the need for simultaneous optimization, but due to their higher cost linked to a more sophisticated structure, they remain limited to those applications really requiring this extra capability, e.g. for submarine links.

Loss minimized fibres (category B1.2 corresponding to Recommendation G.654) have been used for some very long links where the attenuation limitation was predominant.

More recently, it has been recognized that for Wavelength Division Multiplexing (WDM) applications, an important parameter is the stability of chromatic dispersion for the different wavelengths used. It is for this reason that a new category of fibre (category B4, corresponding to Recommendation G.655) called non-zero dispersion fibre (NZDSF) has been generated. Compared to a DSF, a NZDSF exhibits a higher value of chromatic dispersion at 1550 nm, but the slope of the curve chromatic dispersion vs. frequency is reduced in the operating window, in order to facilitate WDM operation.

Standardization of this new category of fibres has been initiated both in the ITU and in the IEC, prompted by decisions of installation in some countries, and some standards values have been agreed upon and published. However, these values provide only a very broad definition and are not sufficient to define precisely a standard product. Several different products, corresponding to different interpretations of the category, can be accommodated within the existing standard and it has not been possible until now to define an agreed set of values leading to a real standard uniform category.

According to IEC Publication 60794-2, which is the most recent publication defining characteristics of fibres, all single-mode fibres have in common several basic parameters such as:

cladding diameter:	125 ± 2 µm
primary coating diameter:	245 ± 10 µm (uncoloured)
	250 ± 15 µm (coloured)

A cladding diameter value of 125 ± 1 µm can be found presently, which provides advantages for splicing purposes.

However, some other parameters differ from one category to another. For example, mode field diameter is:

8.6 to 9.5 µm at 1310 nm for B1.1

7.8 to 8.5 µm at 1550 nm for B2

6 µm at 1310 nm and 7 µm at 1550 nm for B3

under consideration for B1.2 and B4.

Transmission parameters such as attenuation coefficient and dispersion characteristics are obviously different for the different categories. B1, B2 and B3 categories are more or less completely defined as B4 remains under consideration for most of its parameters.

2.1.3 Optical fibre implementation

Optical fibres are normally not used alone but are incorporated in a cable. The purpose of the cable is to protect fibres against mechanical and environmental aggression, but sometimes the cabling and installation processes may impair fibre characteristics if not done properly. Furthermore a single length may not be sufficient and several lengths may need to be spliced together, which requires further consideration of both fibre and cable parameters.

The ITU-T has already published a series of Recommendations on the subject. The most recent of these are :

[L.12] (07/92) – Optical fibre joints

[L.13] (07/92) – Sheath joints and organizers of optical fibre cables in the outside plant

[L.14] (07/92) – Measurement method to determine the tensile performance of optical fibre cables under load

[L.15] (03/93) – Optical local distribution networks – Factors to be considered for their construction

[L.17] (06/95) – Implementation of connecting customers into the public switched telephone network (PSTN) via optical fibres

[L.17 App.1] (02/97)	–	Implementation of connecting customers into the public switched telephone netorks (PSTN) via optical fibres, Appendix 1: Examples of possible applications
[L.20] (10/96)	–	Creation of a fire security code for telecommunication facilities
[L.22] (10/96)	–	Fire protection
[L.23] (10/96)	–	Fire extinction – Classification and location of fire extinguishing installations and equipment on premises
[L.25] (10/96)	–	Optical fibre cable network maintenance
[L.26] (10/96)	–	Optical fibre cables for aerial application
[L.27] (10/96)	–	Method for estimating the concentration of hydrogen in optical fibre cables
[L.28] (10/96)	–	External additional protection for marinized terrestrial cables
[L.29] (10/96)	–	As-laid report and maintenance/repair log for marinized terrestrial cable installation
[L.30] (10/96)	–	Markers on marinized terrestrial cables
[L.34] (10/98)	–	Installation of Optical Fibre Ground Wire (OPGW) cable
[L.35] (10/98)	–	Installation of optical fibre cables in the access network

2.1.3.1 Fibre protection

The first level of protection is provided by the primary coating.

The selection of the primary coating is of paramount importance for insuring the reliability of the fibre. A good primary coating should protect the fibre against mechanical stress, abrasion, etc., as well as environmental and chemical effects such as humidity for instance. It should not create micro-bending (that can happen if it is not applied concentric to the silica material), should make fibre handling easy and should be easily removable (for splicing purposes).

Furthermore, the primary coating in general also provides identification means through a colour code. It should be adapted to this function and the coloured fibber should keep the same characteristics as an uncoloured one. Compatibility between primary coating and the colours used should be demonstrated after production and also for the operating lifetime of the fibre. Stability of colour identification is also needed, as well as compatibility with all the constitutive materials of the cable.

All these requirements show the high level of technology required for the development and implementation of a good primary coating.

In practice, it is not possible to continuously maintain fibres free of stress during their operating lifetime. Some level of stress can be applied, either continuously or for a short time period. It is therefore necessary to consider this situation and to evaluate the consequences on the fibres.

One obvious consequence, when stress is applied to the fibre exceeding some level which is function of cable structure, is a degradation of transmission characteristics, in particular the attenuation coefficient. This issue is therefore quite easy to understand by submitting the cable to appropriate tests.

Another consequence, which cannot be detected immediately but is also very important, is a reduction of expected fibre lifetime, if the fibre is submitted to a stress exceeding some value as described hereafter. This point requires more attention as the result is not directly measurable.

The ability of a fibre to survive mechanical constraints is given by the knowledge of the stress corrosion susceptibility parameters. Definitions and measurement methods for static and dynamic parameters (n_s and n_d respectively), which are both used in practice, are given in IEC Publication 60793-1-3. The lifetime of a fibre under operational conditions is linked to its value of n_s and n_d: the bigger n_s and n_d are, the longer is the expected fibre lifetime for given conditions.

The purpose of the cable is to reduce the level of stress transfer to the fibre itself, due to environment during the operational life of the system, but the above rule related to the n values is always valid: it means that, for a given external stress, if different cable structures transfer differently stress to fibre, the less it transfers, the better it is. But, for a given level of stress on fibre, any increase on the n value will increase the fibre lifetime.

Values of $n_s \geq 20$ and $n_d \geq 25$, shown presently by many products can insure, under normal conditions and based on commonly accepted theoretical calculations, a satisfactory minimum lifetime. Higher values will provide a longer expected lifetime.

The second level of protection is provided by the cable itself. Parameters for optical cables and the way to measure them are given in IEC Publication 60794-1 Optical fibre cables – Part 1: Generic specification.

In particular, the lifetime aspect, as mentioned above, is linked to the absence of micro-bending on the fibre which can be characterized by a thermal cycling test (see IEC Publication 60794-1-2).

The evaluation of the cable behaviour for stress parameters such as tension, crush, impact, bending, torsion, needs also to be done, according to operational requirements. The same reference as above (IEC Publication 60794-1-2) applies.

2.1.3.2 Installation procedures

It is important to keep unaltered, by using convenient installation procedures, the characteristics of optical fibre cables.

Useful information can be found in the informative Annex C of IEC Publication 60794-1-1. The introduction of this document summarizes its main points as follows:

Optical fibre cables are designed so that normal installation practices and equipment can be used wherever possible. They do, however, generally have a strain limit rather lower than metallic conductor cables and in some circumstances, special care and arrangements may be needed to ensure successful installation.

It is important to pay particular attention to the cable manufacturer's recommendations and stated physical limitations, and not exceed the given cable tensile load rating for a particular cable. Damage caused by overloading during installation may not be immediately apparent but can lead to failure later in its service life.

2.1.3.3 Splicing

For the purpose of optical cable splicing, two types of characteristics need to be considered : those related to the fibre and those related to the cable structure.

Concerning the fibre, splicing losses increase with higher tolerances. A reduction of tolerances on cladding diameter to ± 1 μm as mentioned above will reduce splice loss.

A similar result can be achieved by reducing the tolerance on mode field diameter, $2w_0$: the specified value is ± 1 μm and it is possible to achieve values of ± 10.5 μm or lower.

Fibre handling for splicing is characterized by fibre curl (see IEC Publication 60793-1-3). The radius of fibre curl, r, has to be as great as possible and there is a general consensus that a minimum of 2 meters is required. Greater values can be found on the market.

The cable structure is also of interest, considering parameters such as modularity, possibility of easy access to fibres (sheath or tube removal), identification of fibres in the cable structure (colour code, stability of colour over time).

In conclusion, it is clear that optical fibre implementation is governed by a large number of parameters. Some parameters of general interest have been reviewed but it is also important to know precisely the conditions applicable for the operation of the network, during its installation and over its operational lifetime, in order to select the most relevant conditions for the protection of fibres.

2.1.4 Protection of Optical Networks

2.1.4.1 Introduction

Survivability is arguably one of the most important factors in the evaluation and design of fibre optic telecommunication networks. This happens as more traffic is carried by the same fibre infrastructure and more telecommunications service customers are likely to be served by bigger central offices. The evolution of transport networks towards a two layer WDM / SDH has increased even further the emphasis on survivability.

2.1.4.2 Network Survivability

Network survivability is the ability of the network to recover traffic in the event of a failure of a network component, such as the complete loss of a transmission link or the failure of a central office. The main objective of network survivability is to guarantee a certain traffic service level agreement.

Traffic survivability enhancement is achieved by the replacement of failed or degraded transport entities. The replacement is normally initiated by the detection of a defect, performance degradation or an external management request.

The main characteristic of the protection mechanisms as standardized in ITU-T Recommendations like G.841 and G.842 is that they are able to recover traffic very quickly, for most of them, the target is to protect the traffic in less than 50 ms. Further, protection also operates autonomously from the network operation centre. Protection makes use of pre-assigned capacity between the nodes. The simplest protection architecture has one dedicated protection entity for each working entity (1 + 1). The most complex architecture has m protection entities shared amongst n working entities (m:n).

The following protection mechanisms were entirely defined in ITU-T recommendations:

2.1.4.2.1 SNC-P/I (Sub-network Connection Protection with Inherent monitoring)

This protection mechanism uses 1 + 1 architecture, which means it needs a spare sub-network connection to protect a working sub-network protection. It is single ended that is both directions of traffic are protected independently. The faults under which SNC-P/I is able to react and protect are those due to hard failures, occurring as equipment failure, optical interface failures and link failures. Radio link short period interruptions initiating Tributary Unit – Alarm Indication Signal (TU AIS) or Tributary Unit – Loss of Power (TU LOP) will also cause SNCP-I to protect. SNC-P/I may be implemented on a VC per VC basis allowing a given transmission facility to transport unprotected traffic and protected traffic.

2.1.4.2.2 SNC-P/N (Sub-network Connection Protection with Non intrusive Monitoring)

This protection mechanism differs from the previous one, only in the fault conditions under which it is able to protect. SNC-P/N protects against the usual hard failures as described for SNC-P/I but it also protects against failures such as human related activities, operating system mismanagement or mis-provisioning. These types of failures could be originated by a wrong matrix connection or opening of connections. Further, SNC-P/N also protects against soft failures due to degradation of optical interfaces or to errors occurring in a radio link exceeding the predetermined threshold value for maximum acceptable BER.

2.1.4.2.3 MS Linear Trail Protection

This is a bulk traffic protection mechanism able to protect against fibre cuts, optical interface failures, as well as against performance degradation of optical interfaces and fibre. MS Linear trail protection supports many types of architectures, which may be 1 + 1 or 1:N. The bandwidth made available for protection may be used to transport low priority traffic when all the working spans are faultless. MS Linear Trail Protection can be also used to protect spans in linear or chained type of networks.

2.1.4.2.4 MS-SPRING (Multiplex Section Shared Protected Ring)

MS-SPRING is also a bulk traffic protection mechanism able to protect against fibre cuts, optical interface failures, as well as against performance degradation of optical interfaces and fibre. However, MS-SPRING is a protection mechanism that needs a network ring physical topology to work. One of the major advantages of MS-SPRING with respect to other protection mechanisms, is its ability to reuse bandwidth capacity which makes it the most efficient protection mechanism available under conditions of uniformly distributed traffic.

MS-SPRINGs in general are considered to be appropriate in areas of the network where the traffic is uniformly distributed.

There is also a variation of the MS-SPRING implementation in G.841 to optimize it for long distance applications, like submarine networks or backbone networks in large countries, where the distance between nodes can be many thousands of kilometres long. During network failures and while maintenance is taking place, the path under protection can be much longer than in normal operation as ring switches take place adjacent to the network failure. This translates into longer voice transit delay which can constitute an impairment to the conversation fluidity over a long distance PSTN network. To limit this degradation of voice quality, the MS-SPRING functionality has been enhanced to what is called MS-SPINGs for long distance applications and it is described in Annex A of ITU-T G.841

2.1.4.2.5 Restoration

Restoration makes use of any capacity available between nodes to recover traffic against network failures. In general the algorithms used for restoration will involve re-routing. When restoration is used some percentage of the transport network capacity will be reserved for re-routing working traffic. As yet, restoration has not been standardized, the several products currently available in the market respond to a number of specific customer specifications.

Restoration is the most efficient traffic availability enhancement technique if the network physical topology is reasonably well meshed. For example, when the network nodes are reachable through at least three or even better more disjoint physical routes, the amount of spare bandwidth devoted to spare capacity of around 33% or less per span would be enough to restore against single span failures. This is in comparison with the 50% of spare capacity required by the ring protection mechanisms explained above.

The centralised operating system has all the knowledge about traffic distribution, spare capacity, and traffic profiling in function of their priority to be recovered. Depending on the nature of the failure, the restoration mechanism can administer the use of working traffic and other low priority traffic while restoring the higher priority traffic. This feature is called traffic pre-emption. Restoration response is slower than protection due to its high versatility in recovering traffic and its bandwidth efficiency. Centralised automatic restoration mechanisms are able to recover traffic in a time range between 5 to 10 secs without incurring unavailability on traffic for the affected circuits.

2.1.4.3 Selection criteria to chose restoration or protection

To implement cost effective network architectures, ring protection mechanisms and mesh restoration mechanisms should be used together. It is important to understand that these are not possible alternatives for implementation of reliable networks, but they complement each other very well due to their differences and thus should be used together. The criteria for choosing one or other needs to take into account the distribution of fibre and stations in the field. Physical node connectivity is defined as the average number of diversely routed links that arrive to a node (central office). This varies significantly depending on the demographic conditions. In general densely populated areas are served by a well connected or "meshed" network infrastructure. Less densely populated areas tend to have less fibre, with network topologies made up of long chains or large rings. Finally for very low population density, radio and very little fibre constitutes a cost effective way to provide telecommunication services.

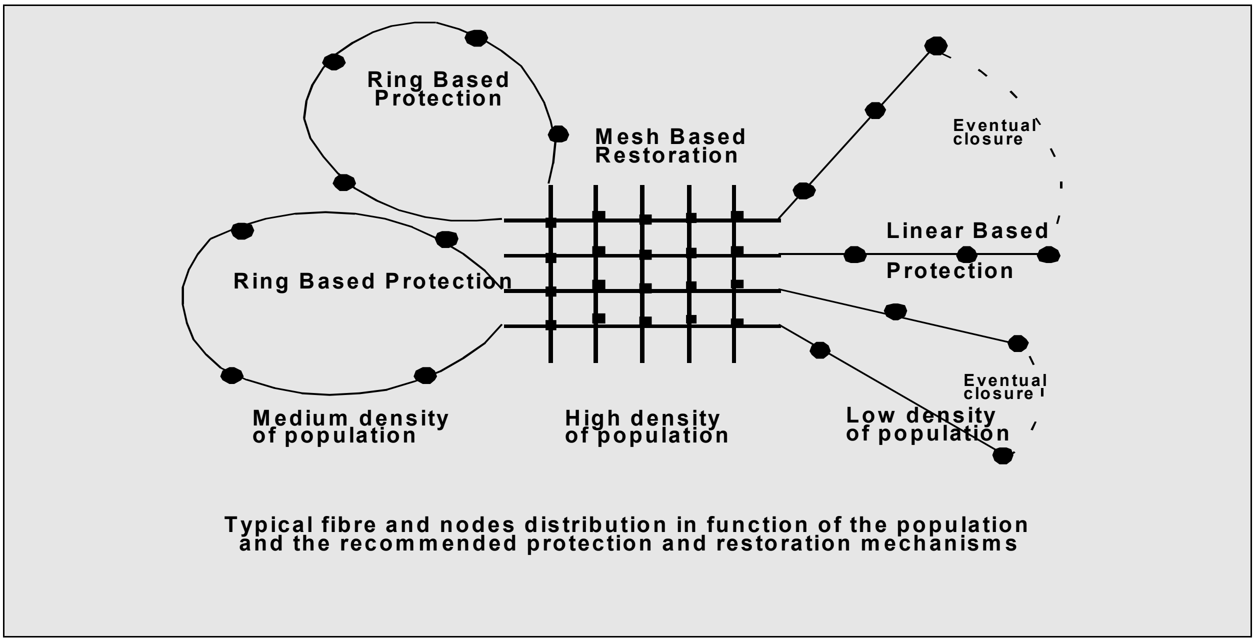
In the cases where the connectivity of the nodes in the network is high enough, for instance higher than 3, the use of meshed restoration can be more appropriate as it is possible to exploit all its virtues. These include its minimum utilisation of spare bandwidth in the network, flexibility of growth and pre-emption.

Most countries have both densely populated areas with a correspondingly well-meshed network to support it, and areas more sparsely populated correspondingly with less fibre connectivity.

Hence, it is possible to state that meshed protection and ring protection can be efficiently used complementing each other to construct reliable high capacity network architectures.

The figure below illustrates this concept which follows the above mentioned criteria of demographic distribution and telecommunication infrastructure required to serve the population.

Figure 2.1.1 – Typical fibre and note distribution



2.1.4.4 Conclusions

A number of economical considerations exist for planning and implementing reliable transport network architectures. High availability transport networks have the merit to support cost effectively a complete range of services and their associated quality. Reliable networks incur in much lower operational expenses on maintenance. As maintenance actions can be deferred and lumped, hence there is much more possibility to plan them better.

Furthermore, the only way to take advantage, safely, of the evolution of the high capacity TDM and WDM systems is to plan for a self-healing network architecture.

Mesh restoration and ring protection mechanisms help construct reliable networks. Studies indicate that ring protection mechanisms are the perfect solution for a poorly interconnected network infrastructure found in medium populated areas.

Restoration with its flexibility and economy of spare bandwidth resources is the best solution to be used with a well inter-connected network infrastructure, which is usually found in densely populated areas. As most countries have areas with high, medium and low density population, a combination of mesh restoration, ring and linear protection respectively constitute the right solution to build reliable transport network architectures for the next century.

2.1.4.5 ITU-T References

For more details in planning optical networks and their protection, the following ITU-T Recommendations are important:

[G.872]; Architecture of optical transport networks

[G.709]; Network Node Interface for the Optical Transport

[G.691]; Optical interfaces for multichannel systems

[G.959.1]; Optical transport networks physical layer interfaces

[G.871]; Framework of optical transport network

[G.798]; Functional characteristics of optical networking equipment

[G.874]; Planned; Management of optical network elements

[G.875]; Planned; Information model for optical network equipment

2.1.5 Abbreviations

BER	Bit Error Rate
DSF	Dispersion Fibre
MS	Multiplex Section
MS-SPRING	Multiplex Section – Shared Protected Ring
NZDSF	Non-Zero Dispersion Fibre
PMD	Polarization Mode Dispertion
PSTN	Public Switched Telephone Network
SDH	Synchronous Digital Hierarchy
SNC-P/I	Sub-network Connection Protection with Inherent monitoring
SNC-P/N	Sub-network Connection Protection with Non-intrusive monitoring
TU AIS	Tributary Unit – Alarm Indication Signal
TU LOS	Tributary Unit – Loss of Power
WDM	Wavelength Division Multiplexing

2.2 Digital radio-relay systems

2.2.1 General

Digital radio-relay systems (DRRS) are used in many applications ranging from transporting telephone and TV signals to carrying a wide variety of modern data signals. Distances bridged may range from less than a kilometre to a continent or beyond. Similarly, the capacity of a digital radio system may be as little as a single DS1 signal (1.54 Mbit/s) or as large as 1,000 Mbit/s. Only a small range of the electromagnetic spectrum is suitable for radio-relay applications and within this range only a limited number of bands are available. The bands are further subdivided into channels that can carry either low capacity or high capacity digital signals.

The growth of digital networks was dictated for more than thirty years by the conversion of voice telephone traffic from analogue to digital. Not until recently has pure data traffic been an important factor. Data traffic is now increasingly generated by voice band modems, ISDN terminals, video conferencing and high quality television terminals and other data sources.

The advantages of digital radio-relay systems include:

– *Low costs:* Radio is cost-effective in comparison with other alternative systems like copper and fibre optic cable. The installation of cable and the cable itself can be very expensive and in urban areas it may be difficult to acquire the necessary rights-of-way.

– *Rapid deployment:* Radio equipment can be easily moved to new sites to meet rapidly evolving network requirements. Infrastructure requirements are small.

– *Ease of maintenance:* Maintenance is limited to the infrequent radio stations along the radio path in contrast to cable system where the entire path is exposed to potential cable breaks.

Spectrum

The ITU conducts periodic international conferences, the World Radiocommunication Conferences or WRCs, where the electromagnetic spectrum is allocated to various users. Regional Radiocommunication Conferences (RRCs) are also held to develop agreements covering the use of the RF spectrum at the regional level. The results are published in the Radio Regulations as the ITU Table of Frequency Allocations, which cover the spectrum from 9 kHz to 400 GHz. In addition to the fixed terrestrial and fixed-satellite services of concern here the spectrum is allocated to many other users like mobile (land, aeronautical, maritime), broadcast, (sound, TV) meteorological, space (operation, research, inter-satellites Earth exploration) radio astronomy, amateur and radio-determination (radar). The Radio Regulations (RR) allocate the spectrum in only the broadest possible terms. Most of the countries use the RR as the basis for their own national frequency tables in which they provide additional details, segregated into government and non-government usage.

The ITU-R Recommendations may not always reflect the latest channel arrangements used in individual countries. New channel initiatives often start in a particular country and are recognized only some time later in an ITU-R Recommendation. If a manufacturer plans to supply radio equipment into a foreign market it is therefore necessary to become familiar with that country's particular developments. A country may also allow a non-standard channel arrangement or open up a government frequency band to non-government use.

In relation with ITU-R Recommendations (F-series), we can take into account the general frequency band arrangements for radio-relay systems:

a) 1.4, 2, 4, 5, L6, U6, 7, 8, 10, 11, 12, 13, 14 and 15 GHz

b) 18, 23, 27, 31, 38 and 55 GHz

Digital radio systems operating below 15 GHz are essential in providing junction and backbone links in the long haul and regional networks. They are also used in remote areas or over difficult terrain and are thus complementary to other transmission systems like optical fibre. Congestion in the bands below 15 GHz makes it often impossible to increase the number of links in an area. That is why the bands above 15 GHz become more and more important. In many countries equipment operating above 15 GHz is being deployed in large numbers for short-range access networks (SDH spurs, LAN, temporary links, and cable protection) and for mobile network infrastructure (e.g. GSM, AMPS, DCS 1800).

Digital channel capacity

The bit rates carried by digital radio-relay systems are the rates standardized in ITU-T Recommendations G.702, G.703 and G.704 for the plesiochronous digital hierarchies and in ITU-T Recommendations G.707, G.708 and G.709 for the synchronous digital hierarchies (SDH or SONET). According to the ITU Recommendations, and depending on the channel bandwidth and the modulation (4 QAM, 16 QAM, 256 QAM, 512 QAM), the capacity of digital radio-relay systems in bit rates f_b includes multiples of DS1 (1.544 Mbit/s) and DS3 (44.736 Mbit/s), E1 (2.048 Mbit/s) and E3 (34.368 Mbit/s), STS1 or Sub-STM1 (51.84 Mbit/s) and STM-1 (155.52 Mbit/s). These are the bit rates delivered to and from the digital radio. Inside the digital radio system the bit rate f_{br} is often about 6% higher (f_{br} = 1.06 f_b) because of the addition of forward error correction (FEC) and the addition of extra overhead bits for internal radio maintenance and to achieve the radio-internal multiplexing of several standard bit streams.

2.2.2 New Digital Microwave (Point-to Point) Radio Technologies

In the remaining part of this section, some of the fields where new technology has appeared over the last few years will be treated in some detail. These technologies are incorporated in new Digital Microwave Radio (DMR) SDH (Synchronous Data Hierarchy) equipment, and will ensure a smooth transition from the currently prevailing Plesiochronous Data Hierarchy (PDH) architecture.

- **FEC (Forward Error Correction) and Coded Modulation Techniques**

In the original and conventional DRRS design, the FEC coding and modulation functions are done independently. Several types of error correction techniques are in use, such as block coding and convolutional coding.

In an FEC block coding scheme, the input data stream to be encoded is divided into *k* information symbols together with redundant parity or check symbols to produce a coded work of *n* symbols which are then modulated and transmitted. An independent block code encoder operating in this way is producing a *(n,k)* or rate *k/n* block code. At the receiver, the demodulated bitstream is first decoded to extract the information bits, which are corrected, if necessary, by the parity or control bits.

In an FEC convolutional coding scheme, parity or control bits are calculated over a span of bits to form a continuous bit stream. Convolutional coding is also described in terms of *n* and *k*. Here, *n* is the number of coded bits that are used to make a certain bit sequence, and *k* is the span of bits that makes the sequence. *k* is called the constraint length. Convolutional coding is usually designed for specific decoders, such as Viterbi decoding, sequential decoding or syndrome decoding.

In coded modulation, the FEC coding and modulation functions are combined by inserting redundant bits in multi-state numbers of the transmitted signal constellation. The most popular coded modulation techniques traditionally used for terrestrial digital microwave radio systems are (1) Block Coded Modulation (BCM), (2) Trellis Coded Modulation (TCM), and (3) Multi-level Coded Modulation (MLCM). Of these techniques, the MLCM technology has seen the most rapid advancement in recent years.

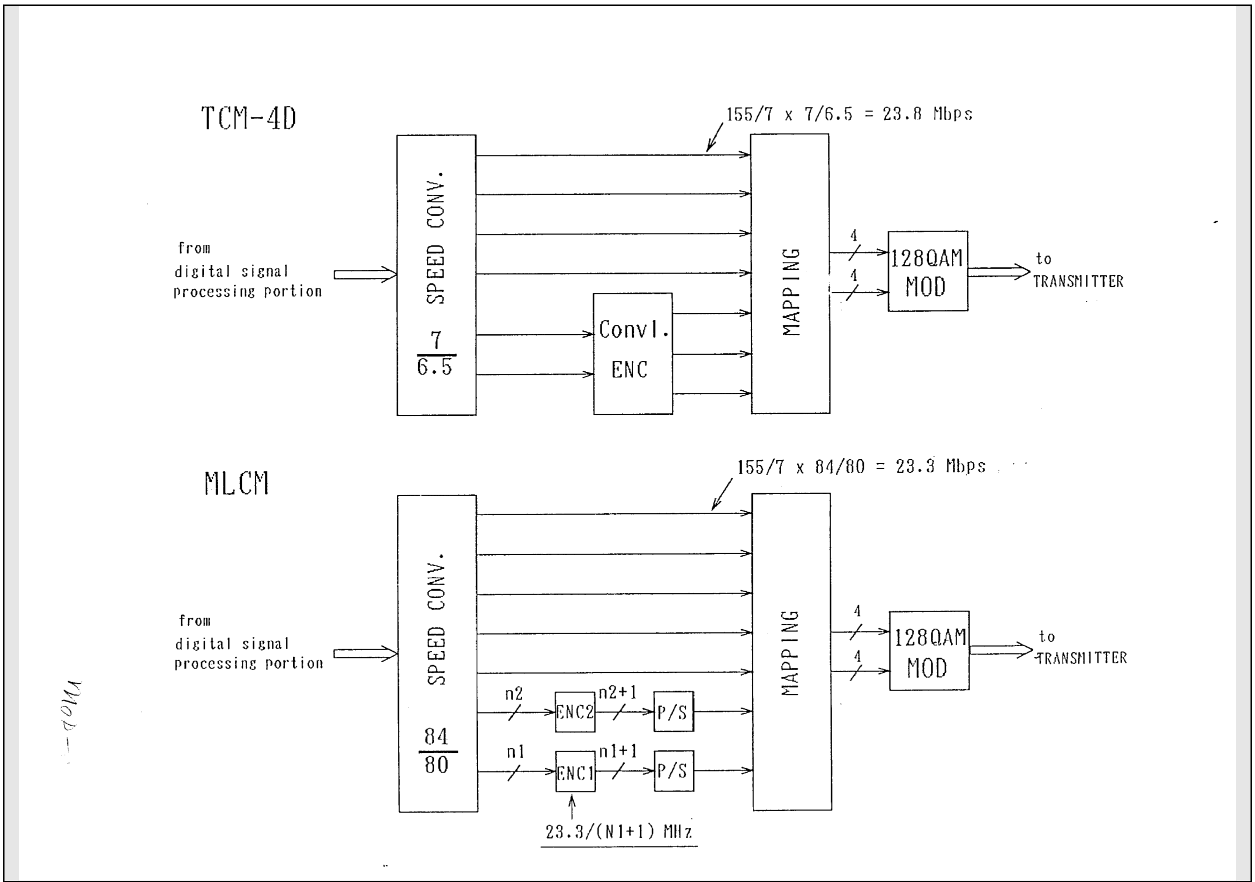
BCM modulation schemes achieve a lower coding gain than TCM schemes. Coding gain (in dB) may be defined as maintaining the same BER on a radio link even if the C/N is reduced. Thus, if a certain modulation scheme has a coding gain of, for example, 3 dB, the C/N can be reduced by 3 dB for the same BER. BCM schemes, however, are simpler than TCM schemes to implement, and may be used in parallel demodulation configurations.

TCM modulation schemes use convolutional coding techniques which require a smaller bandwidth for a given transmission rate and BER probability. The demodulator used for a TCM scheme is often implemented with Viterbi algorithms in a Maximum Likelihood Sequence Estimation (MLSE) circuit. The complexity of TCM schemes can be quite high, but they may have a high degree of flexibility. The characteristics of TCM schemes are such that their coding gain on a non-linear channel is greater than on a linear channel. This advantage of TCM will reduce residual BER for high complexity modulation schemes.

In MLCM each modulation level is considered to be a separate communications channel, and different FEC coding schemes may be applied to different levels. Furthermore, one MLCM scheme is not restricted to one coding scheme. For example, some levels may use block codes while other levels may employ convolutional coding. This means flexibility in selecting coding rates since they can be selected separately for individual levels.

Figure 2.2.1. shows the basic techniques for TCM and MLCM schemes. In the MLCM case, the incoming serial bitstream is speed converted and changed from serial to parallel mode. Encoder 1 will, for example, apply FEC bits to the Least Significant Bit (LSB) at a rate R = 3/4 (i.e. a total of four bits are transmitted for every three information bits), and Encoder 2 may apply FEC bits to the second LSB at a rate of R = 11/12 (12 transmitted bits for every 11 information bits). After this encoding process, the encoder outputs are parallel-to-serial converted and combined with the remaining bit levels in a mapping circuit before being sent to the 128 QAM modulator. Because of the lowering of the operational speed of the coding and decoding processes, a more reliable and robust circuit design is realized.

Figure 2.2.1 – Basic concepts of TCM and MLCM techniques



In the example of Figure 2.2.1 the total MLCM coding redundancy is only about 5% (80/84). This has the advantage that an additional 2 Mbit/s channel can be transmitted (so-called Wayside channel). This advantage is absent in TCM, where for example in 4-dimensional TCM the coding redundancy is 8% and therefore cannot sustain a Wayside channel.

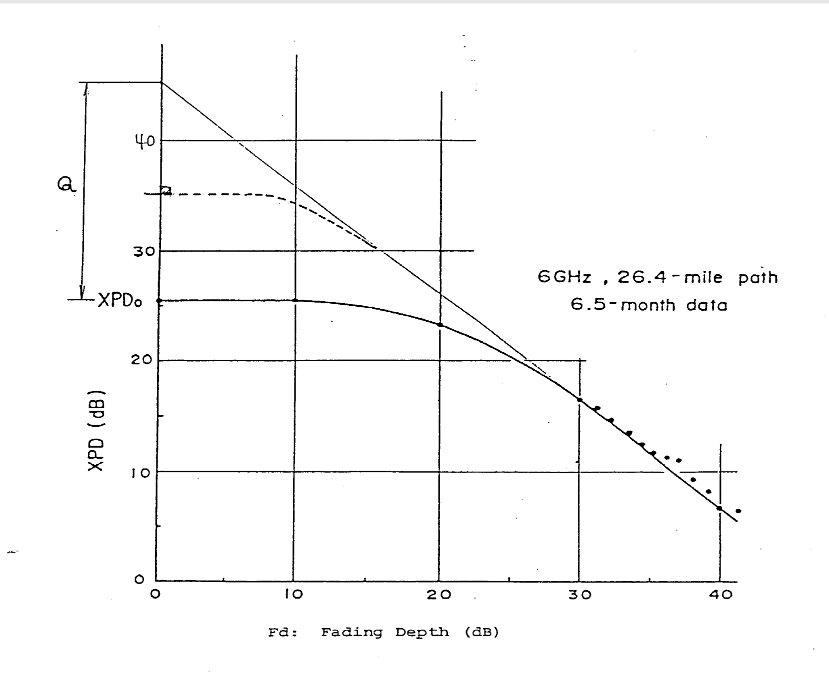
- **Cross Polarization Interference Canceller (XPIC) Techniques**

The frequency bands used in digital microwave radio systems (as well as in other radio communications systems) are limited resources, and therefore should be utilized in the most effective way possible. Increasing the level of modulation as described in the previous section is one way to increase the capacity of a given system. Another way to increase the use of a given frequency band is to use it twice in the same system by transmitting RF carriers with different polarizations.

Co-channel dual polarization equipment has been in operation in commercial satellite systems for many years, but only more recently has it been introduced into terrestrial microwave systems. The reason for this is that satellite systems are rarely operated below antenna elevation angles of 5 degrees and propagation fading is not a critical issue. Terrestrial microwave systems, however, are inherently susceptible to severe fading because they are transmitting parallel to the surface of the earth, and cross-polarization discrimination (XPD), i.e. interference of signals between the two polarizations, is highly affected by multipath fading RF carriers.

Figure 2.2.2 shows the expected XPD vs. fading depths. The data shown in this figure are for an antenna designed for high cross-polarization discrimination. As shown, under non-fading conditions an XPDo (XPD in normal condition) of 35 dB may be expected.

Figure 2.2.2 – Expected cross-polarization vs. fading depth



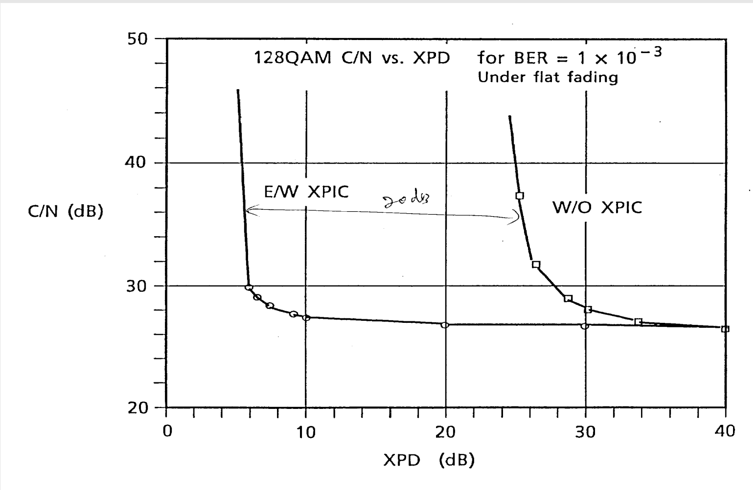
In order to maintain the XPD between the two polarizations even under fading conditions, equipment developers have turned their attention to designing antennas with increased XPD characteristics about the antenna boresight and cross-polarization interference cancellers (XPIC) utilizing transversal equalizing technology. Since terrestrial systems often transmit several RF carriers simultaneously, adjacent channel interference should be addressed at the same time by proper filter designs with sharp cut-off characteristics. A 256 QAM system, for example, will improve the band efficiency from approximately 7 bits/Hz to 14 bits/Hz.

The XPIC circuit takes samples of the interfering signals and applies them to the desired signals for cancellation of the interference. The cancellation may be done at either RF, IF or at the baseband level. The complexity of the XPIC circuitry increases with multilevel coded modulation schemes, and systems that employ 256 QAM and 512 QAM modulation states call for very sophisticated XPIC designs. The benefits to be gained, however, are very reliable communications links even in the face of severe fading.

Due to controlled imperfections in the antennas, some interference will be expected in the IF signals. The vertically polarized carrier will have interference from the horizontally polarized carrier, and vice versa. The amount of interference is dependent on the antenna XPD and the amount of fading. In the absence of fading, the interference is minimum and acceptable.

Figure 2.2.3 shows the improvement that can be expected in XPD performance by using cross polarization interference canceller circuits. This figure indicates measured results for a system employing 128 QAM modulation techniques at a constant BER rate of 1×10^{-3} under flat fading conditions. The expected improvement is approximately 20 dB.

Figure 2.2.3 – Measured XPD improvement by using XPIC equipment



- **Automatic Transmission Power Control (ATPC)**

Inclusion of automatic transmission power control (ATPC) circuitry in DMR systems will reduce interference particularly between channels on the same route and between channels transmitted from the same system site. By using ATPC, several advantages are accomplished, including: (1) reduction in angular separation between adjacent radial routes, (2) reduction in the distant interference between hops which reuse the same frequency, (3) reduction of interference between adjacent digital and analogue channels that share the same frequency band.

Further equipment related advantages by using ATPC are realized, including: (4) reduction in the DC power consumed by the RF amplifier. This reduction may be quite significant (up to 40%), (5) improvement in the residual BER performance (typical 1×10^{-13}),

The dynamic range of ATPC circuits is typically between –12 dB and + 2 dB in controlled steps of 1 dB with fading tracking speed of 100 dB/sec, and the threshold level of the received signal may be preset in a range between –50 dBm and –70 dBm.

- **Equalizers for Counteracting Multipath Fading**

Multipath fading is of major concern on most DMR links implemented in the world. Traditionally, the adverse effects of multipath fading have been mitigated by installing redundant equipment operating in space diversity or frequency diversity configurations.

Another way to combat the effects of multipath fading is to use adaptive equalizers in time or frequency domains. Recent technological advances have improved the performance of these equalizer circuits quite significantly, which is particularly important for DMR systems that use multi level coded modulation schemes.

Adaptive equalizers may be implemented at the IF or baseband levels, and the most common circuits in use are the so-called decision feedback equalizers (DFE) and linear transversal equalizers.

Fading effects on a digital radio link are often depicted as a system “signature”, which is a static indication of the susceptibility of equipment to a two ray model of the multipath channel in both minimum phase and non-minimum phase conditions. It is often used for the purpose of equipment comparison.

Due to the dynamic nature of multipath fading, dynamic tests should be performed on the equipment. This is often done by using a dynamic simulator that can simulate the time sequences of multipath fading. Such simulation tests are used for optimization of synchronization and equalizer coefficient adapter circuits. Figure 2.2.4. shows the measured signature of a 64 QAM 155 Mbit/s digital radio system with decision feed-back circuits included. The ability of equalization against minimum phase fading of the linear equalizer is the same as that against non-minimum phase fading. In contrast to that, the adaptive time domain equalizer shown in Figure 2.2.4 equalizes minimum phase fading completely and it is possible to equalize severe fading in which the reflective wave becomes bigger than the main signal.

2.2.3 A modern SDH system utilising new technologies[1]

Long-haul digital microwave radio systems have been recommended by the ITU-T in 1988 to adopt Synchronous Digital Hierarchy (SDH) configuration, in an effort to synchronize equipment over wider geographical areas. SDH systems are thus taking over from Plesiochronous Digital Hierarchy (PDH) systems which have been in operation for a number of years. SDH equipment should be designed and implemented with a smooth transition from PDH in mind.

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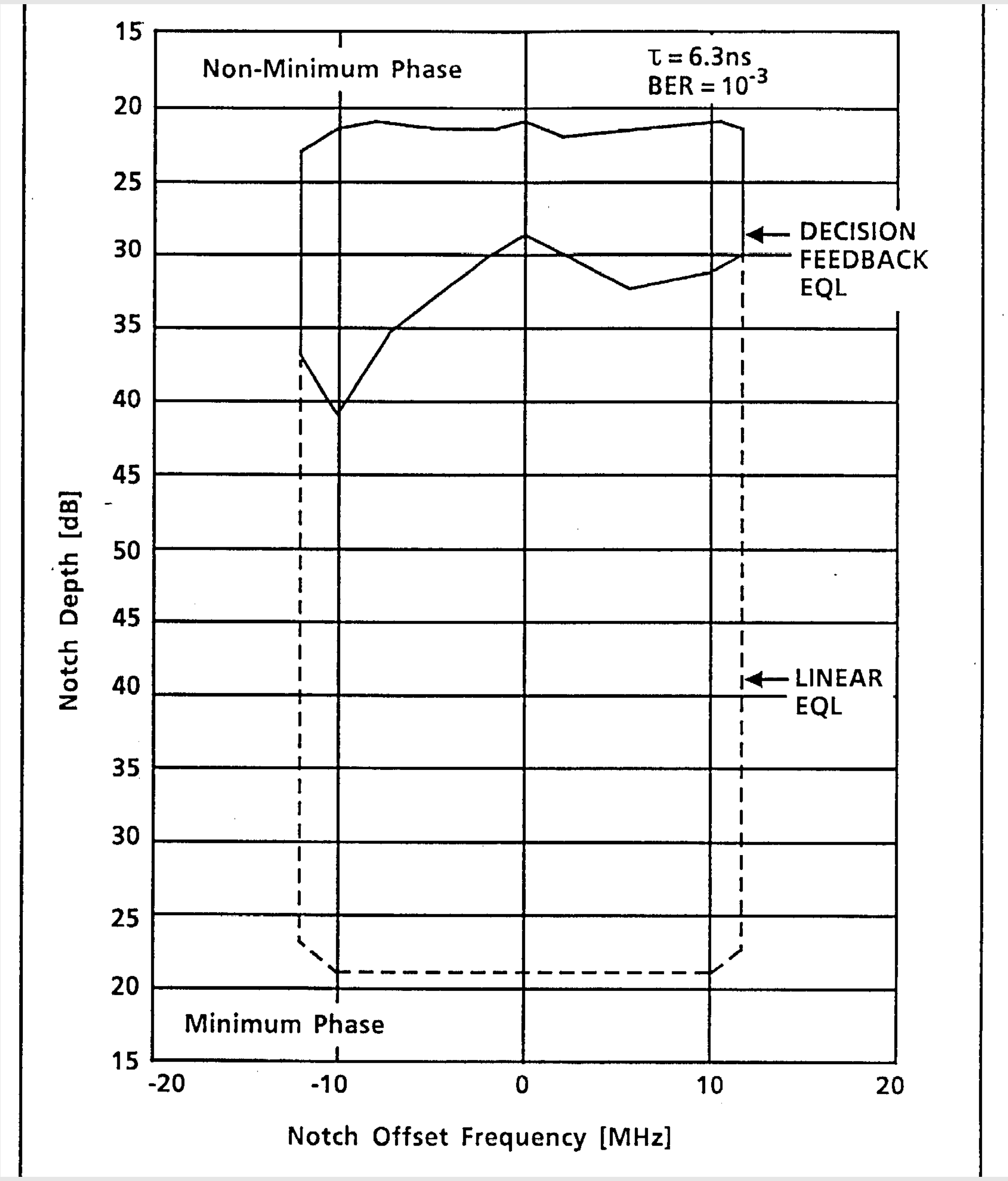
[1] See more about SDH principles in Chapter 6.

Figure 2.2.4 – Measured signature of 64 QAM 155 Mbit/s system with DFE



SDH systems generally operate in frequency bands between 4 and 13 GHz using 64 or 128 Quadrature Amplitude Modulation (QAM) and have transmission capacity of Synchronous Transmit Module 1 (STM-1). They can be accommodated in existing RF frequency channel slots vacated by PDH systems.

A modern SDH digital microwave radio system should include features that make it easy to operate and maintain, and which make the most effective use of the available RF bandwidth. New technological advancements as outlined in this book should be included to the extent possible. Features that should be implemented include:

- Conformity with the latest international standards, specifically ITU-R, ITU-T and ETS standards.
- Protection against fading such as Frequency Diversity (FD) and/or Space Diversity (SD) switching. Furthermore, the system should be traffic capacity expandable preferably by simply adding equipment modules.
- The system gain should be high by using receivers with low noise figures. In systems utilizing FD and/or SD protection switching, the switching should not result in any drastic reduction in Bit Error Rate (BER) performance.
- Multi-Level Coding Modulation (MLCM) should be used in order to ensure improved effectiveness and performance. Coding gain should be optimized and coding redundancy should be minimized to enable insertion of additional 2.048 or 1.544 Mbit/s service channel(s) in the Radio Frame Complementary Overhead (RFCOH) field.
- Automatic Transmit Power Control (ATPC) should be considered in order to reduce interference to neighbouring systems, improvement of the residual BER performance, alleviation of fading problems, and reduction in power consumption.
- Decision Feedback Equalizers (DFE) or other suitable equalization schemes should be included in order to reduce inband dispersive amplitude and delay distortion caused by multi-path fading.
- Two different external synchronization clocks should be available for redundancy purposes. This should be in addition to an internal oscillator clock.
- The SDH system should be capable of being integrated into existing PDH baseband networks. Upgrading of the interface to STM-1 transmission should be software selectable.
- The Section Overhead (SOH) should preferably be terminated in two directions, i.e. not only for the radio section but also for the electrical or optical line section which is terminated at the radio station.
- The system should be equipped with an electrical or an optical STM-1 interface for each channel. The optical interface should be available for two or more different cable lengths: for intra-office and long haul inter-office applications.
- In addition to the STM-1 or 140 Mbit/s main traffic stream, several (Digital Service Channels (DSC), and depending on the MLCM scheme used, one or two additional 2.048 or 1.544 Mbit/s service channel(s) should be dropped and inserted at every terminal and repeater radio station. These channels should be transported in the RFCOH.
- Extensive performance monitoring and control functions should be built into the equipment. Such parameters should be in conformance with ITU-T G.784. Furthermore, a count should be kept of the number of protection switch-over operations, and an account of the failed time for individual channels. Such functions should preferably be performed from a Network Management System (NMS).
- In order to enhance the reliability of the equipment and extend the MTBF figure, extensive use of LSI Microwave Integrated Circuits (MIC) and Hybrid IC (HIC) should be implemented. Furthermore, low-power high-speed Complementary Metal Oxide Semiconductor (CMOS) LSI should be used wherever possible in order to make the equipment more compact and reduce the power consumption.
- Equipment should be installed in standard racks with room for expansion, and should be able to operate over a wide range of environmental conditions.
- The equipment should be flexible and should be reconfigurable.

- Orderwire communications should preferably be implemented with a priority orderwire and an omnibus orderwire using SOH bytes E2 and E1, respectively. The orderwire channels should provide voice communications throughout the entire SDH radio network using selective calling facilities by means of Dual Tone Multifrequency (DTMF) signalling.
- Management features should be designed in accordance with, amongst others, ITU-T Recommendations of the M.3000 series and G.784. This enables interconnection to TMN-based Network Management System (NMS) from various vendors by means of the so-called Q3 interface (see more information see Fascicle 2, Chapter 5 *Network and Service Management*).

2.2.4 Multiple Access Radio Systems

Despite the fact that people all over the world are crowding into metropolitan or urban areas in ever increasing numbers, the major part of the world's population is still living in rural areas and to a large extent has the same aspirations and requirements for telecommunications services as their counterparts in the big cities. Such rural population is often spread thinly over vast geographical areas.

National telecommunication organizations that have been charged with providing communications to all areas of a country have traditionally not had many options available to them, and particularly not for providing high-grade services like television and high-speed data communications. Often, the only choice has been a pair of copper wires.

With the advent of high-powered satellites in recent years, VSAT (Very Small Aperture Terminal) systems can be used for providing high-speed data and television services to rural areas. Such systems may be established quickly and rather inexpensively per terminal. However, VSAT systems are normally looked upon as low capacity facilities, and covering a vast thinly populated area with VSAT terminals may be an expensive proposition. Widespread inexpensive hand-held satellite telephone sets operating with Low or Medium Earth Orbit (LEO/MEO) satellites are still a few years away, and in any case the initial costs and running and operating costs of such devices may put them out of reach for many rural areas (see more details in subchapter 2.4 *Satellite Systems* of this Fascicle).

As an alternative for rural communications, multiple access subscriber radio communication systems may be considered as an attractive solution. One such system employing modern digital technology may be implemented with up to approximately 1000 subscriber lines spread over a radius of about 1000 km from one base station. That is more than 3000 square kilometres per subscriber, and in terms of economy and services provided, such a system may compete with VSAT systems and LEO/MEO hand-held satellite telephone sets.

2.2.4.1 System Architecture

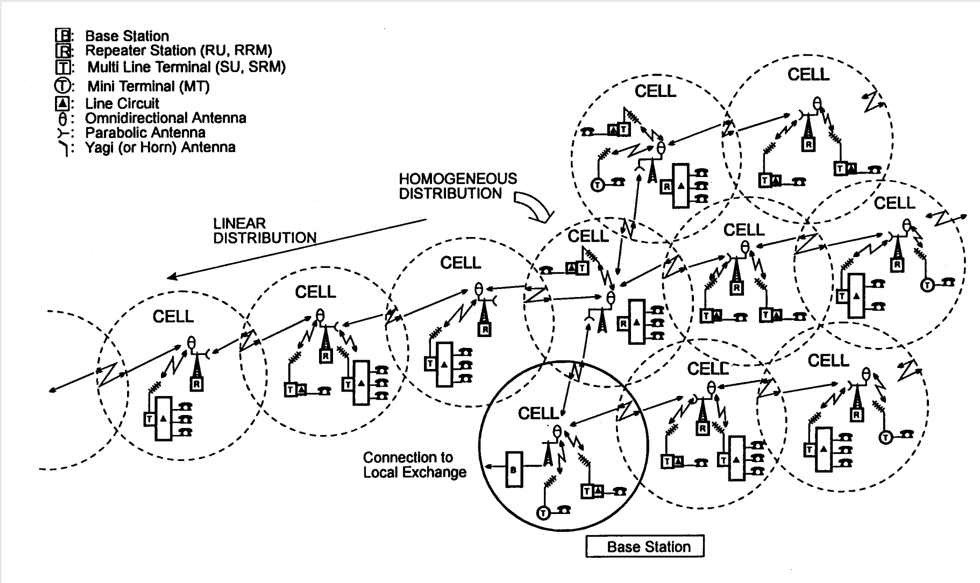
A subscriber radio system consists of three major elements, namely the base station, repeater stations and terminal (subscriber) stations. A system may be implemented in a homogeneous configuration (typical of an urban system), or in a linear configuration (typical of a rural system). These systems operate in the 1.5-2.4 GHz frequency bands and typically use TDM (Time Division Multiplexing) technology with 60 time slots for 1024 subscribers in the downward direction (i.e. from the base station to the subscribers), and TDMA (Time Division Multiple Access) technology in the upward direction. Voice channels are encoded at 64 kbit/s Pulse Code Modulation (PCM). Such a system is often referred to as a Digital Radio Multiple Access Subscriber Systems (DRMASS). In newer systems the voice coding is done at 32 kbit/s Adaptive Differential PCM and the number of time slots is approximately doubled.

A subscriber radio system is constructed with cells in a honeycomb configuration, not unlike cellular radio systems for mobile users. Figure 2.2.5 shows the basic concept of subscriber radio system and Figure 2.2.6 is an example of a typical cell arrangement and frequency allocations for a 14-channel DRMASS system. The base station is typically located at the same site as a metropolitan or urban telephone exchange for easy interface of the DRMASS circuits into the national telephone network. The interface may be either digital (for example at 2 Mbit/s) or 2-wire analogue circuits. Each cell in the DRMASS system has radio transmit and receive equipment for passing calls on to the next cell or for communicating with subscribers under its jurisdiction.

The subscriber equipment in a DRMASS system includes multiline subscriber units for locations with several subscribers grouped together, and mini terminals for locations with a single or two subscribers in one location.

In technology recently developed, a cell in the DRMASS system may be included for mobile and/or fixed users. Such a cell is referred to as a Digital Cordless Telephone System (DCTS) cell and is used for communicating with cordless telephone sets or small fixed terminals at 32 kbit/s ADPCM (Adaptive Differential Pulse Code Modulation).

Figure 2.2.5 – Digital radio multiple access subscriber system basic concept



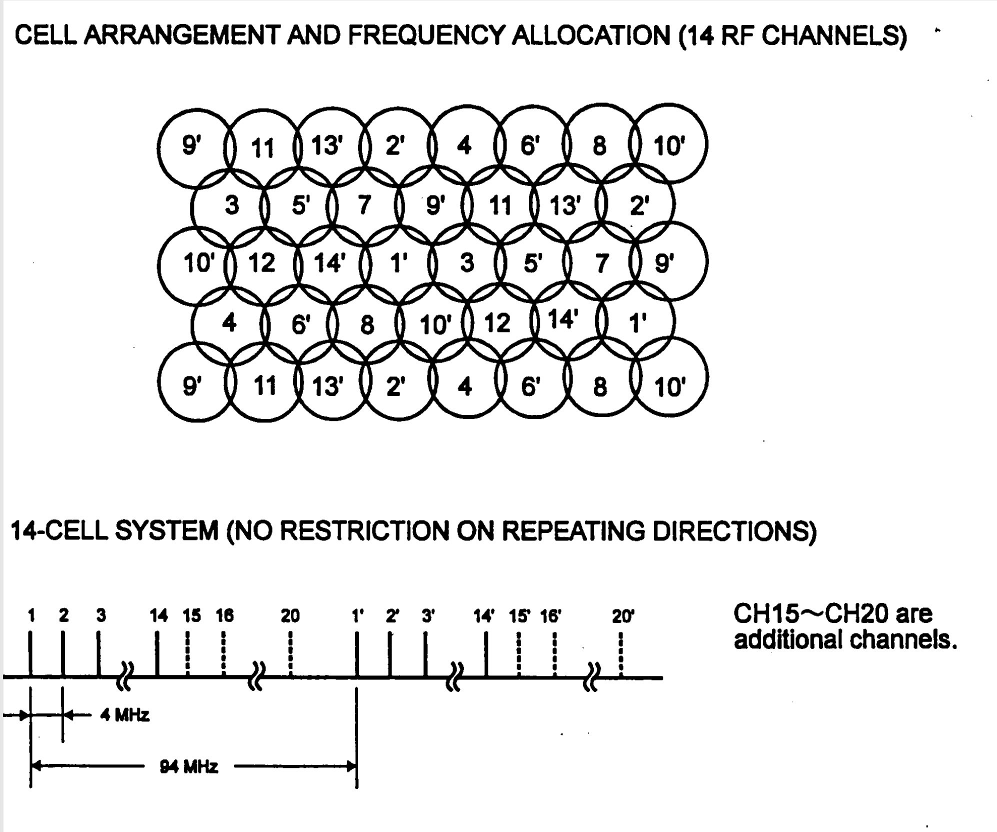
In DRMASS systems each cell repeater station can cover an area up to a radius of approximately 45 km, and a DCTS cell can communicate with fixed terminals at a radius of approximately 3 km.

A DRMASS system provides its services by concentrating a large number of subscriber lines into available time slots, and the system is normally transparent to signalling to and from the subscribers.

The telephone channels coming from the local telephone exchange are concentrated and converted to digital signals and transmitted down to the nearest repeater and onwards to other repeaters if applicable. Each repeater will communicate with its subscriber from an omni directional antenna. One time slot among the 60 slots is automatically assigned to the subscriber making a call. For the upward transmission from the subscriber via a repeater station to the base station, the subscriber is assigned a time slot and his transmission is in burst mode (TDMA) without any risk of overlapping with other subscribers. Only one pair of frequencies is used for upward and downward transmissions.

In Figure 2.2.7 the overall frame format for one frame is shown together with the various fields of information in the frame. Figure 2.2.8 shows a typical format for one subscriber time slot channel (so-called V-channel).

Figure 2.2.6 – Example of cell arrangement and frequency allocations for 14 channels



The radio frame consists of 60 voice channels (V CH), one control channel (C CH), one supervisory channel (SV CH), one orderwire channel (OW CH), one telegraph channel (TELEX CH), one acquisition and local maintenance channel (ACQ/LOCAL CH), and a frame synchronization pattern (F, F'). The frame period is 4 msec, and the frame pattern is allocated 7 bits in the leading section of both the C CH and the ACQ/LOCAL CH channels. The first part of the leading section is a fixed word of 5 bits and the remaining 2 bits is a variable word (ID NO.). These words are used to discriminate between the routes, and as frame identification. Identifications of every 16 frames are done by a test module. The polarity of the ACQ/LOCAL CH and the C CH is inverted in the fixed word and this polarity alternates at intervals of 2 msec.

Frame synchronization uses a one bit shift hunting method, and synchronizing protection is achieved by 3 continuous 'yes'/4 continuous 'no' pulses.

Each V CH, OW CH and SV CH has 152 bits for both upward and downward signal flows and consists of the actual data (for example 64K voice) and control information. In the upward signal flow, two guard time slots are provided at the beginning and the end of the frame. The C CH has 128 bits and is used for sequencing call connections. The ACQ/LOCAL CH has 280 bits. When in the acquisition mode, it is used for delay adjustments at the initial system line-up. When in local maintenance mode it is used for delay fine adjustments. The TELEX CH also has 152 bits and may be used for low data rates (up to 19.2 kbit/s) or up to 40 telex channels. The V CH has the same structure when used for telex or data service. Modulation is normally QPSK, and therefore 2 bits of PCM data are transmitted per symbol. Hence, for a 4 ms frame, 256 bits of PCM data are transmitted (64 kbit/sec) per V CH.

Figure 2.2.7 – Frame format in a DRMASS TDM/TDMA system

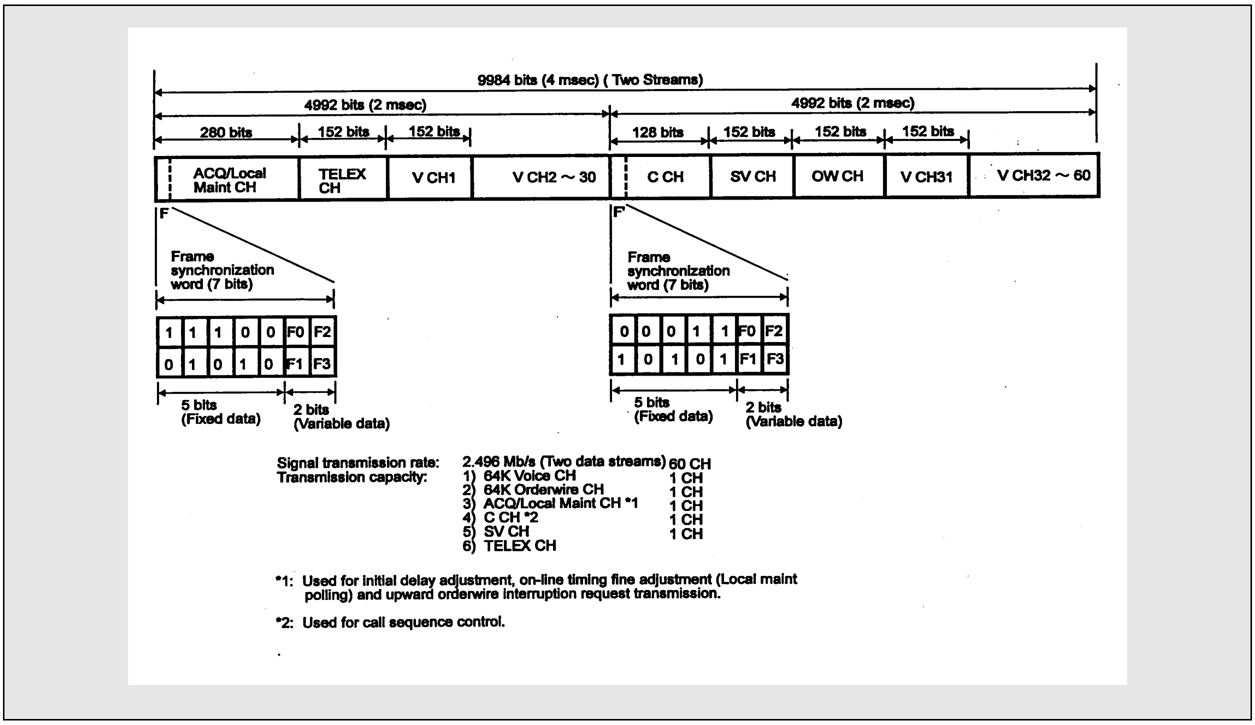
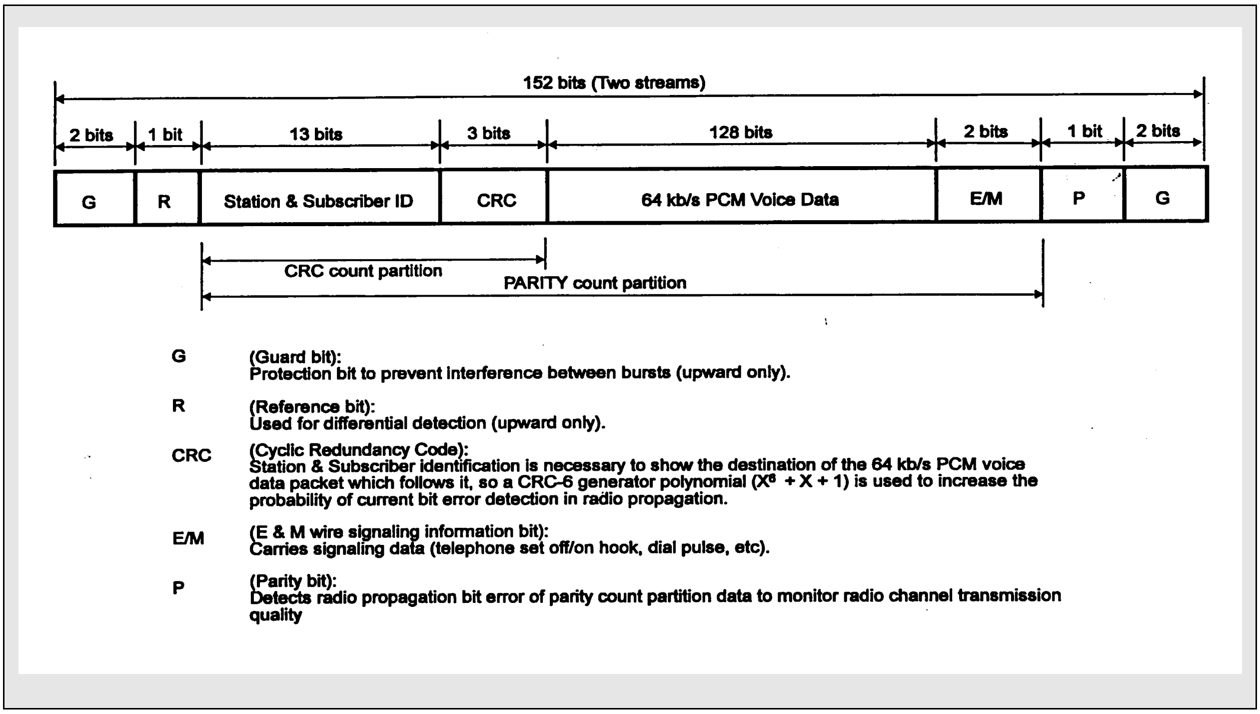


Figure 2.2.8 – Frame format of Subscriber Time Slot



- **Services Provided and Grade of Service**

A modern multiple access subscriber radio system operates at a high technology level and should be able to provide at least the following services:

- Ordinary telephone service
- Coin telephone service
- Data transmission, typically up to 384 kbit/s which, with modern software, is adequate for providing PC-to-PC medium quality video conferencing
- Telex service
- Dedicated (pre-assigned) telephone and data channels
- 4-wire communications with E&M signalling
- Intra-call service. This refers to calls between subscribers using the sam terminal station.
- Emergency priority call service. If used, non-priority calls may be dropped.
- Priority automatic call-back service. If a subscriber has this priority, his call may be placed in a queue if the system is busy. He will be called back automatically ahead of ordinary call-backs when a channel is available.
- Ordinary automatic call-back service
- Call forwarding on busy/no response/unconditional status
- Call waiting
- Three-way calling
- Supervisory Control and Data Acquisition (SCADA) operations
- 4-wire DAMA (Demand Assigned Multiple Access) service
- ISDN (Integrated Services Digital Network) service

Since a DRMASS system utilizes the concentrator concept, the maximum number of subscribers that can be handled in a system is a function of the average traffic generated by each subscriber, the desired blocking probability, and the number of time slots.

Based on experience, the average business hour traffic call rate of a DRMASS system is typically set at 0.09 erlang/line. This is for an example design where coin telephone sets (0.3 erlang/line) is assumed to use 10% of the system capacity, business communications (0.1 erlang/line) constitute 30%, and residential subscriber lines (0.05 erlang/line) are 60% of the system capacity. The blocking probability (i.e. the ratio of busy calls to the total number of calls in a certain time period) is typically set at 0.01 or 1%. Based on these assumptions, a DRMASS system with 60 time slots will be suitable for 512 subscribers. However, by implementing intra-calling in the system, the number of subscriber lines can be doubled to 1024.

- **The Base Station**

Since in most cases a subscriber radio system has to be interfaced into a national switched telephone network, the most practical location of the base station is at an existing exchange that already may have a radio tower that can be used, and may have enough power generating AC/DC equipment to power the base station. However, where this is not practical, part of the base station equipment (including the RF modules) may be located anywhere, and the remaining equipment in the telephone exchange area as long as the two sites can be interfaced by a 2 Mbit/s digital connection or wireless communications links.

The major equipment elements in the base station are the following:

- One Base Station Controller Module (BSCM)
- One TDM Controller Unit (TCU) including RF transmitter, receiver and antenna.

– One or several Loop Open End Modules (LOEM)
– One or several Data Units (DU)
– A computerized supervisory and control system for operations and maintenance purposes

If for some reason it is not practical to install all the base station equipment at the telephone exchange site, the radio tower with antenna and the TCU module may be installed in a separate location and connected to the BSCM by two 2 Mbit/s links.

The functions of the base station communications equipment are described as follows:

Base Station Controller Module (BSCM)

This module is acting as a concentrator and provides 2-wire interface with up to 1024 subscriber line circuits. It will concentrate 1024 telephone lines and convert them into 60 time slots in two 2.048 Mbit/s TDM data streams. The concentrator contains processor units which, under stored program control, perform call sequencing and execute remote supervision of the complete system.

Loop Open End Modules (LOEM) and Data Units (DU)

The LOEM module(s) provide(s) the interface to the local exchange by plug-in cards. Typically, two types of cards are available, namely one for 8 ordinary telephone lines and one for 4 coin telephone lines. The type and quantity required depend on the LC (Line Circuit) configuration in the subscriber and repeater stations.

The DU module contains plug-in cards for data channel interfaces with the telephone exchange. The configuration of these cards depends on the overall DRMASS system configuration and contains interfaces for data ports, telex channels, and 4-wire E&M signalling.

TDM Controller Unit (TCU)

The TCU converts the two 2.048 Mbit/s data streams from the concentrator into two 2.496 Mbit/s radio TDM signals for the “downward" transmit path and reverses this for the “upward” receive path. Orderwire and supervisory/maintenance signals are multiplexed/demultiplexed in the TCU. Duplicated (working and standby) configuration is normally implemented for higher reliability. In addition, an unprotected version is available for low-cost applications. The TCU can be located remotely from the BSCM concentrator equipment via two 2.048 Mbit/s communications links with ITU-T G.703 interfaces.

The radio or TDM part of the TCU performs the following functions:

– It converts the two 2.048 Mbit/s data streams from the concentrator to packet data for radio communications, and multiplexes the control, supervisory and maintenance signals onto the packet data. The data is modulated and transmitted to the nearest repeater station and onwards to the next repeater station(s) and subscriber stations, as applicable.
– It receives the burst mode TDMA RF signals from the nearest repeater station, regenerates the packet data, removes the control, supervisory and maintenance information from the baseband data and converts the signals into two 2.048 Mbit/s data streams and sends them to the concentrator.

Within the TCU, a main processor controls the TCU side of the data transfer between the TCU and the BSCM for control of the radio protocol. The frame converter converts the baseband 2.048 Mbit/s PCM frame to the radio frame 2.496 Mbit/s format (upward), and from 2.496 Mbit/s to 2.048 Mbit/s (downward). It also multiplexes and demultiplexes the radio frames, and supervises radio transmission errors.

The 60 time slots are transmitted in RF carriers using QPSK modulation. Transmitter power is typically about 1 watt from a transistor amplifier, and the receiver has a noise figure of about 3 dB. It will maintain its operation with acceptable BER rates down to about – 85 dBm. Depending on the system configuration, omni directional antennas (gain = 10 dB), or up to 4-meter parabolic antennas (gain = 32-37 dB) are typically used.

- **The Repeater Stations**

The repeater stations serve two main purposes. Firstly, a certain repeater station will communicate with subscribers within its coverage area. Typically, a total of 256 subscribers may be located in this coverage area. Secondly, it will pass on a call to the next repeater if the call is not intended for one of its own subscribers. There is no real limitation to the number of repeaters in a system, but due to practical considerations one DRMASS system is rarely implemented with more than 23 repeaters and an overall distance of 1080 km.

Figure 2.2.9 shows a block diagram of a segment of a typical DRMASS system configuration. 2 repeater stations and 2 subscriber stations are shown (one mini-terminal with 2 telephone lines served by the base station, and one multi-line terminal which may contain up to 16 telephone lines, served by a repeater station).

The repeater station receives the “downward" signal stream transmitted from the base station (or preceding repeater station). After regenerating the signal, the repeater transmits it on a different radio frequency, not only to the subscriber units within its service area, but also to neighbouring repeaters for further repeating. The “upward” signal is transmitted towards the base station in the reverse direction.

The multiple use of repeaters allows the service area to be expanded to virtually any distance without any noticeable degradation in the signal quality because digital transmission with regeneration of signals is used.

A repeater station may be implemented as an indoor unit or an outdoor unit. The major equipment blocks are the repeater unit itself for receiving, regeneration, and retransmission of RF signals, and one or more drop-out unit. One drop-out unit will typically serve 64 subscribers in an outdoor-type repeater station. Indoor repeaters may contain up to 4 drop-out units for service to 256 subscribers.

- **The Subscriber Stations**

A subscriber station may be a multi-line terminal or a mini-terminal. A multi-line subscriber station is composed of a SU (Subscriber Unit) and a DOU (Drop-out Unit) combination for a capacity of up to 64 subscribers for out-door type repeaters. Alternatively, a SRM (Subscriber Rack Mount) and a DORM (Drop-out Rack Mount) combination may be used to provide up to 256 subscribers capacity (indoor type repeater). The mini-terminal for indoor installation is for small subscriber capacities (typically one telephone set and one coin telephone line).

Due to the low power consumption of the subscriber units achieved by the use of LSI technology, solar-powered operation is made possible for these terminals. The terminal equipment and solar control unit including batteries can be accommodated in small cabinets suitable for installation either on a wall of an existing building or on an antenna pole in an outdoor environment.

A subscriber station performs the following functions:

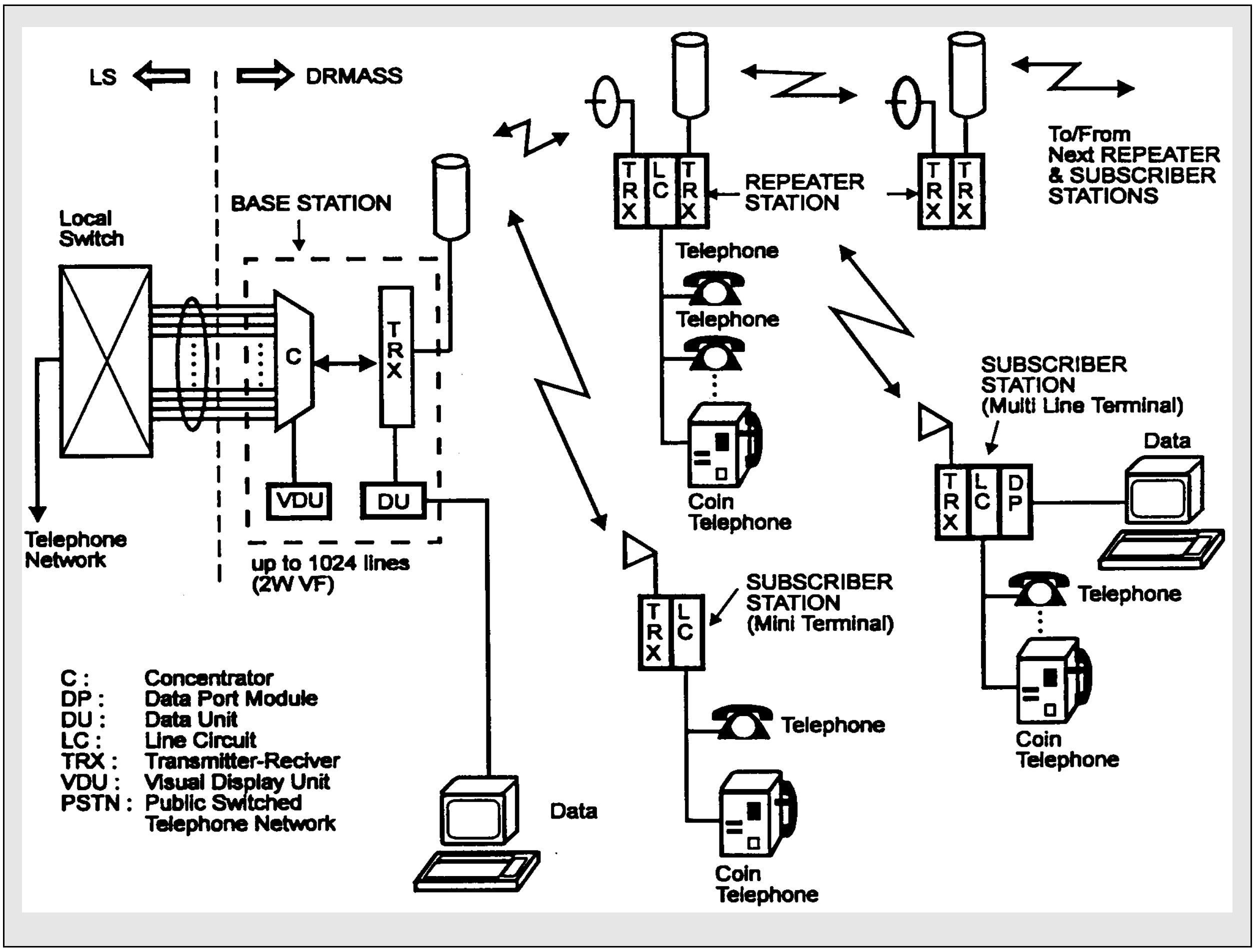
– It receives the RF transmission from the base or repeater station and regenerates the 2-wire analogue telephone signal information and relays this information to the proper subscriber line.

– It detects incoming call requests from the subscriber line and converts the 2-wire analogue telephone signals to RF bursts which are transmitted to the repeater or base station.

Data port cards may be substituted for voice cards in order to provide data communications channels.

The mini-terminal is suitable for locations where only one or two subscriber lines are required. These lines may be a combination of services such as two telephone sets, two coin telephone lines, one telephone and one coin line, one telephone line and one data line, or one coin telephone line and one data line.

Figure 2.2.9 – Example segment of DRMASS system configuration



- **Digital Cordless Telephone System (DCTS) Cell Stations**[2]

Small economical mobile or fixed telephony terminals with limited range may be integrated into a subscriber radio system. These terminals are using the most advanced digital cordless telephony technology and may be operated within a range of 3-5 km from a DCTS cell station for fixed terminals and up to about 1 km for portable handsets. Figure 2.2.10 shows a conceptual drawing of a cell station with all electronic equipment including power supply mounted on a pole.

Portable terminals will typically use a short whip antenna, and fixed installations will use a small pole-mounted Yagi antenna. The terminals use 32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM) and dynamic channel assignment is employed in the system. The DCTS cell stations also operate in the 1.9 GHz band. A maximum of 52 subscribers can access one cell station by using TDMA techniques with a traffic loading of 0.1 erlang and a 1% blocking rate. Voiceband data and fax service may be used up to a speed of 7.2 kbit/s.

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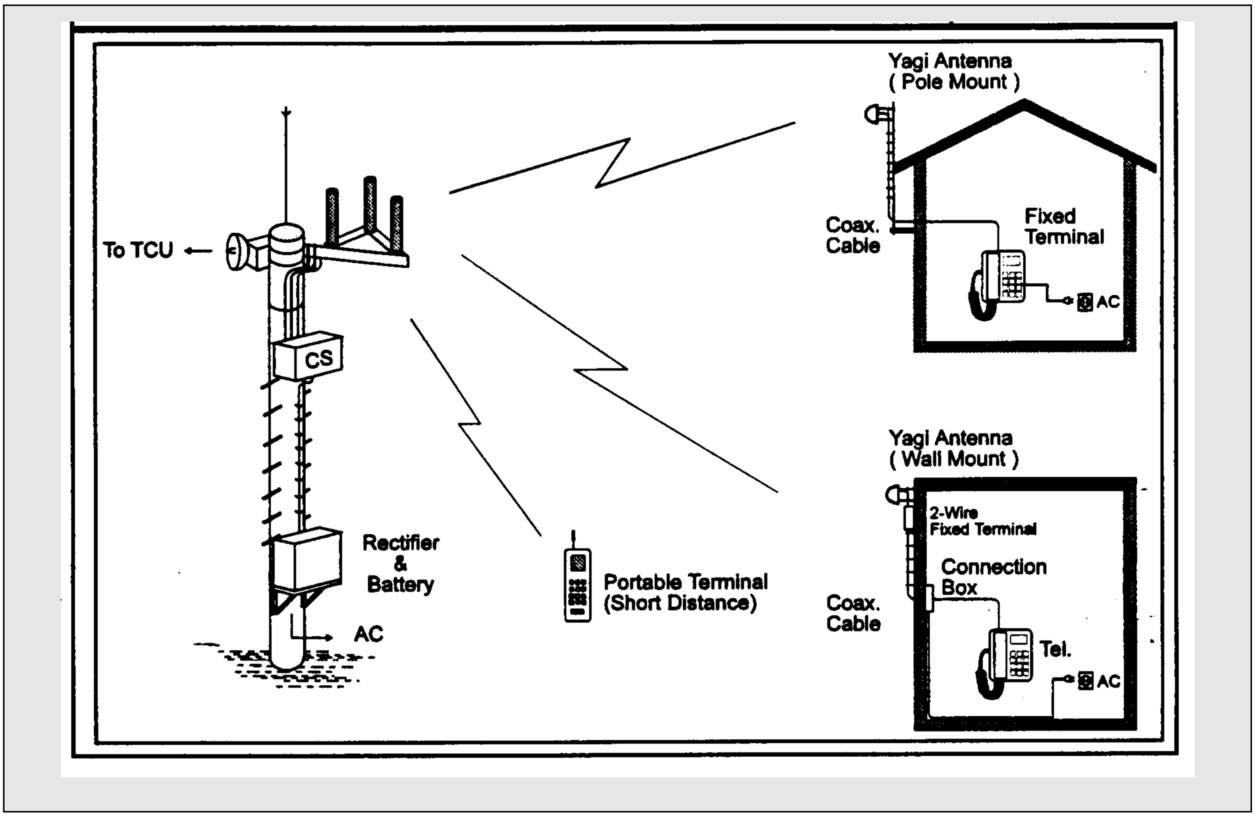
2 See also Fascicle 2, Chapter 4.

- **Centralized Operations and Maintenance Control**

In trend with operating and maintaining modern communications systems, multiple access subscriber radio systems will have automatic centralized operations and maintenance facilities. This is in particular important for a system employed for rural communications where all equipment is operating unattended often in inhospitable environments, and where technical expertise is unavailable and long in arriving if required.

Figure 2.2.10 – DCTS cell station (CS)



A centralized operations and maintenance system will normally be co-located with the base station equipment and will contain facilities for overall system equipment status monitoring, maintenance and testing. Specifically, from one central position, one operator should be able to monitor and control functions within the following areas:

– Alarm status and reporting.

– Subscriber line testing (insulation resistance/capacitance), and test calls to specific subscribers.

– Traffic statistics reporting.

– System reconfiguration.

All above functions should be performed remotely and controlled from the base station. If such a system is implemented together with the multiple access subscriber radio system, it is possible to operate a 1024 subscriber system spread over a 1080 km radius from the base station with one maintenance engineer and one technician, provided the equipment is designed to modern MTBF (Mean Time Between Failures) requirements.

2.2.5 Digital Radio Networks

a) Long haul digital radio systems

Long haul, high capacity digital radio systems can be competitive with optical fibre, especially in difficult terrain, like mountains, across lakes or rivers, in urban areas with expensive or unavailable rights-of-way, or where speed of deployment is important. Also, digital long haul radio systems can re-use a vast system of towers and buildings, previously used by analogue radio systems employed in national backbone telecommunications networks. Channel capacities employed by these systems are the maximum that can be achieved with high-state QAM modulation schemes. A large number of channels may be operated on the same route by using an $n + 1$ switching system. Because of the long distances involved the lower frequency bands of 4 and 6 GHz have been widely used in many countries.

b) Short haul digital radio systems

Today digital radio systems find a major application in all those cases where cable laying would be difficult, costly and time consuming. Digital radio systems can be deployed very rapidly, especially if small unobtrusive antennas can be located on existing buildings or towers and distances are relatively short. Equipment operating at frequencies above 15 GHz meet these requirements very well because they are small, rugged and economical. The explosive growth of cellular networks for mobile radio has generated a large market for these millimetre wave radios. They interconnect cell sites with Mobile Switching Centres (MSC) using capacities that are relatively low, ranging from 1-DS1 to 4-DS1 or 1-E1 to 4-E1. Because of the low capacity, protection is sometimes omitted or otherwise provided by automatic hot standby switching. And since hop lengths are often short there is no need for frequency or space diversity protection against multipath fading. Rain attenuation is the predominant cause of signal outage, which can be kept within ITU requirements by the use of high system gain radios and short hops.

Where larger hop lengths are required, digital radios operating below 15 GHz have to be used. The 2 GHz band was used extensively in the United States of America for low capacity connections by power and gas companies. This band has now been allocated to various PCN uses and the existing radio services can be redeployed in the Lower 6 GHz (L6), Upper 6 GHz (U6) and 11 GHz bands which have been opened up for low capacity digital applications. These bands are not affected by rain fading and therefore provide large distance capabilities. Space and frequency diversity systems are used here as countermeasures for multipath fading with frequency diversity systems normally operated in the form of an $n + 1$ protection switching system. In the interest of spectrum conservation, the FCC in the United States of America demands that the number n grows to at least 3 within three years after first deployment on a route. This rules out permanent 1 + 1 frequency diversity systems. Other countries may have other rules or no such requirements.

c) Radio Local Area Networks (RLAN)

Today's computer-based business activity largely depends on communication infrastructure provided by *local area networks* (LANs). LANs have to be extended in line with the increase of terminal users, and therefore designed to handle bursty traffic in order to efficiently share computer resources. However a wired LAN has many constraints in the aspects of cost, maintenance and installation in particular for networks with complicated architecture

Many types of RLAN exist including those already put into service or to be realized in the near future. Table 2.2.1 summarizes typical examples of RLANs using frequency bands above 1 GHz and having maximum data rate higher than 1 Mbit/s. Information on wireless local loop technologies is available in Fascicle 2, Chapter 4.2.2.1.

2.2.6 ITU Publications

ITU-R Handbook on Digital Radio-Relay Systems issued in the 1996 represents a comprehensive summary of basic principles, design parameters and current practices for design and engineering of digital radio-relay systems. It is issued in all ITU languages. Another useful ITU Handbook is Rural Telecommunications, Volume I – Radio Systems in Rural Areas, 1994.

Table 2.2.1 – Examples of RLAN characteristics

Frequency band	Modulation and/or access scheme	Data rate (typical)	Application	Range[1] (typical)
403-470 MHz 806-869 MHz 946 MHz	4-level FSK	19.2 kbit/s	ARDIS[2] subscriber equipment	ARDIS service area
850 MHz (cellular)	FSK	14.4 kbit/s 9.6 kbit/s (Fax)	Personal communication via cellular phone	Cellular phone service area
902-928 MHz	Frequency hopping (FSK)	64 kbit/s to 500 kbit/s	Point-to-point data link campus and private networks	4 km
	Direct sequence	2 Mbit/s 215 kbit/s to 1.0 Mbit/s	Portable LAN Ethernet LANs	250 m 100 to 1 000 m
	CDMA/TDMA spread spectrum	1.536 Mbit/s line rate	Personal communication networks	450 to 5 000 m^2
	Direct sequence with 1.5 MHz frequency channel selection	60 kbit/s	Bar-code reading	120 to 210 m
	Direct sequence PSK trellis code	5.7 Mbit/s	Ethernet LAN (IEEE 802.3)	80 m
2.4 to 2.4835 GHz 2.4 to 2.485 GHz (transceiver to hub) 5.745 to 5.830 GHz (hub to transceiver)	CDMA, direct sequence frequency hopping Direct sequence 16 PSK trellis code	1 Mbit/s (Approx.) 5.7 Mbit/s	– Ethernet LAN (IEEE 802.3)	– 80 m
5.2 GHz	GMSK (BT = 0.4)	24 Mbit/s Raw data rate	High performance RLANs (HIPERLANS)	50 m
17.2 GHz	Specification in progress	Specification in progress	High performance RLANs (HIPERLANS)	Specification in progress
18.8 GHz 19.2 GHz	TDMA-TDD 4-FSK	15 Mbit/s	Ethernet LAN	40 m (maximum)
19.5 GHz	TDMA-TDD 4-FSK	25 Mbit/s	Ethernet LAN	40 m (maximum)

1 The range of operation of RLAN systems may vary greatly depending on data rate, frequency, RF power, antenna and the propagation environment.

2 ARDIS: advanced radio data information service.

2.2.7 Abbreviations

ACQ/LOCAL CH	Acquisition and Local Maintenance Channel
ADPCM	Adaptive Differential Pulse Code Modulation
AMPS	Advanced Mobile Phone System
ARDIS	Advanced Radio Data Information Service
ATPC	Automatic Transmission Power Control
BCM	Block Coded Modulation
BER	Bit Error Rate
BSCM	Base Station Controller Module
C CH	Control Channel
CDMA	Code Division Multiple Access
CMOS	Complementary Metal Oxide Semiconductor
CRC	Cyclic Redundancy Check
DAMA	Demand Assigned Multiple Access
DCS	Digital Cellular System
DCTS	Digital Cordless Telephone System
DFE	Decision Feedback Equaliser
DMR	Digital Microwave Relay
DOU	Drop-out Unit
DORM	Drop-out Rack Mount
DRMASS	Digital Radio Multiple Access Subscriber System
DRRS	Digital Radio Relay System
DTMF	Dual Tone Multi-Frequency
DU	Data Unit
ETS	European Telecommunication Standard
FCC	Federal Communications Commission
FD	Frequency Diversity
HIC	Hybrid Integrated Circuit
IC	Integrated Circuit
IF	Intermediate Frequency
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
ITU-R	International Telecommunications Union – Radiocommunications
LAN	Local Area Network
LC	Line Circuit
LEO	Low Earth Orbit
LOEM	Loop Open End Module
LSI	Large Scale Integration
MEO	Medium Earth Orbit
MIC	Microwave Integrated Circuits
MLCM	Multi-level Coded Modulation

MLSE	Maximum Likelihood Sequence Estimation
MSC	Mobile Switching Centre
MTBF	Mean Time Between Failure
NMS	Network Management System
OCH	Orderwire Channel
PCM	Pulse Code Modulation
PCN	Personal Communication Network
PDH	Plesiochronous Digital Hierarchy
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
RFCOH	Radio Frame Complementary Overhead
RLAN	Radio Local Area Network
RR	Radio Regulations
RRC	Regional Radiocommunication Conference
SCADA	Supervisory Control and Data Acquisition
SD	Space Diversity
SDH	Synchronous Digital Hierarchy
SOH	Section Overhead
SONET	Synchronous Optical Network
SRM	Subscriber Rack Mount
STM-1	Synchronous Transmission Module 1
SU	Subscriber Unit
SV CH	Supervisory Channel
TCM	Trellis Coded Modulation
TCU	TDM Controller Unit
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TELEX CH	Telex Channel
V CH	Voice Channel
VSAT	Very Small Aperture Terminal
XPD	Cross Polarisation Discrimination
XPDo	Cross Polarisation Discrimination in normal condition
XPIC	Cross Polarisation Interference Canceller

2.3 Mobile communication systems

2.3.1 Introduction

Mobile radio systems provide their users with opportunities to travel freely within the service area being able to communicate with any telephone, fax, data modem, and electronic mail subscriber anywhere in the world; to determine their own positions; to track precious cargo; to improve the management of fleets of vehicles and the distribution of goods; to improve traffic safety; and to provide vital communication links during emergencies, search and rescue operations, etc. These *tieless (wireless, cordless)* communications, exchanges of information, determination of position, course, and distance travelled etc. are made possible by the unique property of the radio to employ an *aerial (antenna)* for radiating and receiving electromagnetic waves. When the user's radio antenna is stationary over a prolonged period of time, the term *fixed radio* is used; a radio transceiver capable of being carried or moved around, but stationary when in operation, is called a *portable radio*; a radio transceiver capable of being carried and used, by a vehicle or by a person on the move, is called *mobile radio*. Individual radio users may communicate directly or via one more intermediaries, which may be *passive radio repeater(s), base station(s), or switch(es)*. When all intermediaries are located on the Earth, the terms *terrestrial radio system* and *radio system* are used; when at least one intermediary is satellite borne, the terms *satellite radio system* and *satellite system* are used. According to the location of a user, the terms *land, maritime, aeronautic, space,* and *deep-space radio systems* are used. The second unique property of all terrestrial and satellite radio systems is that they all share the same natural resource – the *airwaves (frequency bands and the space).*

Recent developments in **microwave monolithic integrated circuit (MMIC), application specific integrated circuit (ASIC)**, analogue/digital signal processing (A/DSP), and battery technology, supported by **computer aided design (CAD)** and robotics in manufacturing allow a viable implementation of miniature radio transceivers. The continuous flux of market forces (excited by the possibilities of a myriad of new services and great profits), international and domestic standards forces (who manage the common natural resource – the airwaves), and technology forces (capable of creating viable products), acted harmoniously and created a broad choice of communications (voice and data), information, and navigation systems, which propelled an explosive growth of mobile radio services for travellers.

The terms fixed (radio) service and mobile (radio) service are defined by the IUT Radio Regulations:

Fixed service: a radiocommunication service between specified fixed points.

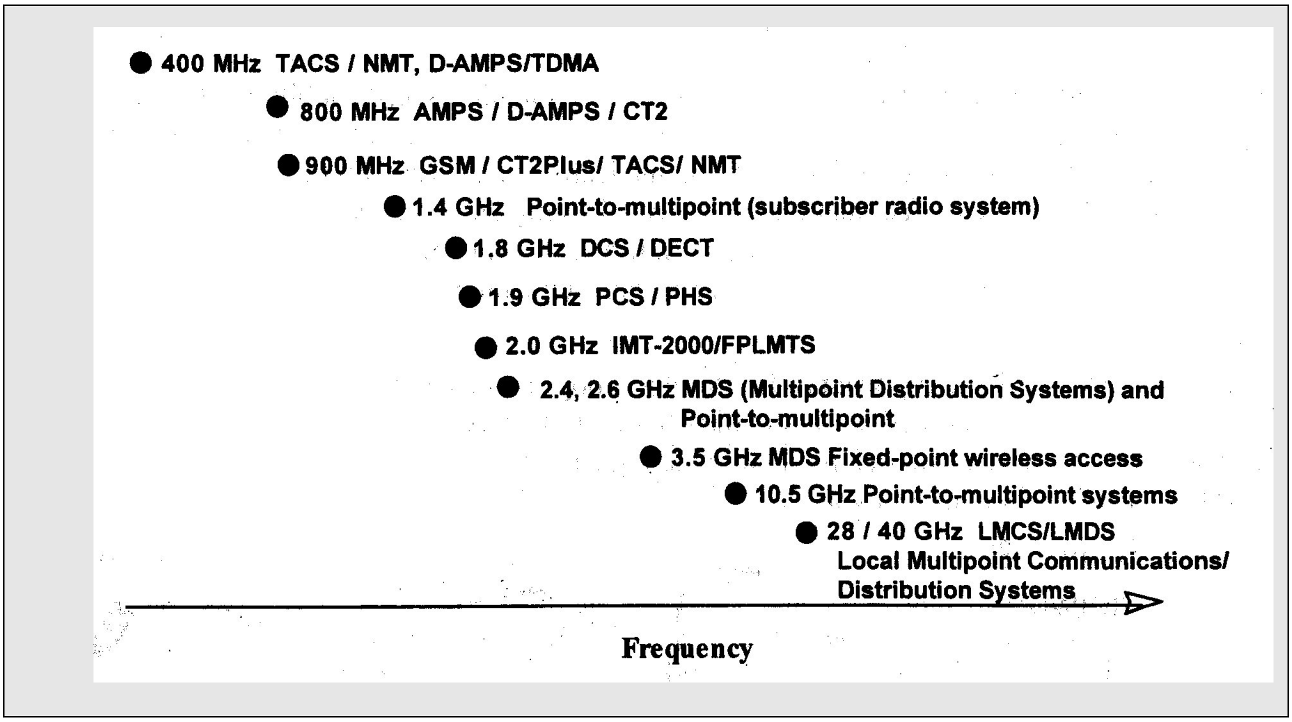
Mobile service: a radiocommunication service between mobile and land stations, or between mobile stations.

Indeed, these radio service definitions form the basis for the allocation of radio spectrum internationally by the ITU as well as domestically within each country. For the most part, the ITU has made joint allocations to the mobile and fixed services in various frequency bands. Traditionally, in some countries a choice has been made between the two services.

2.3.2 Spectrum

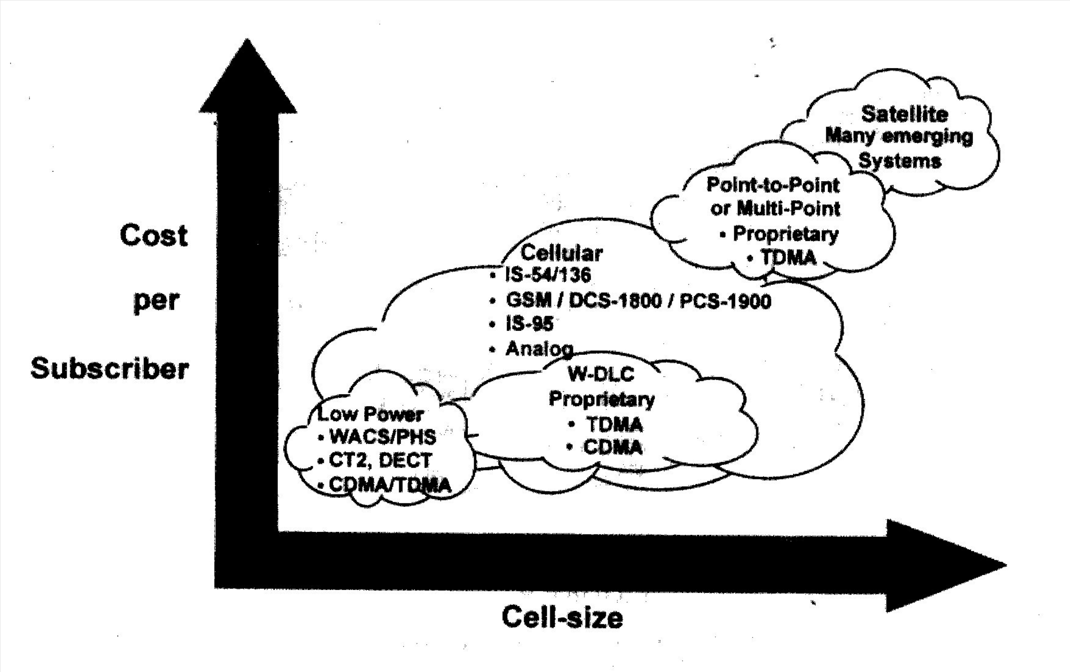
Key input parameters to the design of a wireless network are teledensity over time and subscriber usage (e.g., Erlangs per subscriber). These variables establish the traffic density (e.g., Erlangs per square kilometre) that the wireless system needs to support. The availability of spectrum, channelization of the particular radio access technology, the extent to which frequencies can be re-used, and cell-size, all influence the traffic carrying capacity of a radio-based system. Wireless access systems, from an RF planning/network viewpoint, are deployed in a similar way to cellular systems, even though the teledensity requirements may be different. Radio spectrum requirements and availability need to be examined. Figure 2.3.1 shows bands of the radio spectrum where fixed wireless access systems operate in some countries. The precise availability of radio spectrum for these applications is subject to local regulations. For example, in addition to the bands 1.4 and 2.4/2.6 GHz, the 3.5 GHz band has been endorsed by CEPT/ETSI, as is the 10.5 GHz band for use in Europe. Also, in Region 2 the 3.4-3.7 GHz band is being considered for wireless access applications in some countries. The applications shown in Figure 2.3.1 for different frequency bands may be known by different names which are related to the traditional use of these bands. However, the common underlying aspect is wireless access, which may take many flavours depending on the signal transfer capabilities offered (e.g., voice, data, image, video), mix of service (e.g., purely fixed, mixed fixed/mobile, nomadic), and range (e.g., use of repeaters).

Figure 2.3.1 – Examples of bands of radio spectrum for wireless access



The choice of technology needs to be related to capacity and cost requirements as it is shown in Figure 2.3.2.

Figure 2.3.2 – Mapping technology onto appropriate requirements



2.3.3 Terrestrial Systems

In a terrestrial mobile radio network, a repeater was usually located at the highest point in the vicinity, offering maximum service area coverage. As the number of users increased, the available frequency spectrum became unable to handle the increased traffic, and a need for frequency reuse arose. The service area was split into many small sub-areas called cells, and the term cellular radio was born. Frequency reuse offers an increased system capacity, whereas the smaller cell size can offer an increased service quality but at the expense of increased complexity of the user's terminal and network infrastructure. The tradeoffs between real estate availability (base stations) and cost, the price of equipment (base and mobile), network complexity, and implementation dynamics dictate the shape and the size of a cellular network.

2.3.3.1 Analogue Cellular Technologies

Analogue cellular modulation is based on the principle of Frequency Modulation (FM). The speech to be transmitted is used to modulate the carrier causing it to be varied in amplitude and frequency. Most analogue cellular systems use Frequency Modulation (FM) for speech and Frequency Shift Keying (FSK) for data. In analogue cellular systems, a traffic channel is usually modulated on one radio frequency.

Table 2.3.1 gives some of the operating analogue cellular radio systems: the North American AMPS, the Japanese land MCS-L1 and MCS-L2, the Nordic NMT-900, the German C450, and the United Kingdom's TACS.

Table 2.3.1 – Comparison of Radio Systems

Parameter	AMPS	MCS-L1 MCS-L2	NMT	C450	TACS
TX freq. MHZ Base Mobile	 869-894 824-849	 870-885 925-940	 935-960 890-915	 461-466 451-456	 935-960 890-915
Multiple Access	FDMA	FDMA	FDMA	FDMA	FDMA
Dulpex Method	FDD	FDD	FDD	FDD	FDD
Channel bw, kHz	30.0	25.0 12.5	12.5	20.0 10.0	25.0
Traffic Channel per RF Channel Total Traffic Channels	1 832	1 600 1200	1 1999	1 222 444	0 1 1000
Voice	**Analogue**	**Analogue**	**Analogue**	**Analogue**	**Analogue**
Sylabic comp.	2:1	2:1	2:1	2:1	2:1
Speech rate kbit/s	–	–	–	–	–
Modulation	PM	PM	PM	PM	PM
Peak dev. KHz	±12	±5	±5	±4	±9.5
Ch. Rate kbit/s	–	–	–	–	–
Control	**Digital**	**Digital**	**Digital**	**Digital**	**Digital**
Modulation	FSK	FSK	FFSK	FSK	FSK
BB waveform	Manch.	Manch.	Manch.	Manch.	Manch.
Peak dev., kHz	±8	±4.5	±3.5	±2.5	±6.4
Ch. Rate, kbit/s	10.0	0.3	1.2	5.3	8.0
Channel coding	BCH	BCH	B 1	BCH	BCH
Base – Mobile	(40.28)	(43.31)	Burst	(15.7)	(40.28)
Mobile-Base	(48.36)	a.(43.31) p.(11.07)	Burst	(15.7)	(48.36)

- **AMPS – Advanced mobile phone system**

The AMPS air interface is currently specified by the American National Standards Institute, Electronic Industries Association (EIA), and Telecommunications Industry Association (TIA). The current version is EIA/TIA-553.

- **Nordic mobile telephone (NMT)**

Standard Nordic systems (NMT) are designed to operate within the 400-470 MHz band. NMT was developed in the late 1970's, and was operational before cellular systems were implemented. This system operates in a manner similar to a cellular system and is used by Iceland, Sweden, Finland, Switzerland and Holland. A modified version is also used by Denmark.

- **Total access communication system (TACS)**

Total Access Communication System (TACS) is a 15 MHz frequency spectrum used in Europe, some parts of Asia, the Middle East and Africa.

Channel spacing	25 kHz
Duplex spacing	45 MHz
Frequency range for channels:	
Base receive/mobile transmit	890.0125-904.9875 MHz
Base transmit/mobile receive	935.0125-949.9875MHz
Total number of voice channels	558
Signalling channels	42

- **Expanded total access communication system (E-TACS)**

E-TACS adds an additional 16 MHz of spectrum to the existing TACS system and is considered an extension of TACS No additional signalling channels are provided; however, 640 voice channels are allocated –320 for Band A and 320 for Band B. Transmit and receive frequencies are separated by 45 MHz.

Channel spacing	25 kHz
Duplex spacing	45 MHz
Frequency range for channels:	
Base receive/mobile transmit	872.0125-887.9875 MHz
Base transmit/mobile receive	917.0125-932.4975 MHz
Total number of new voice channels	640
Total number of voice channels (TACS and E-TACS combined)	1198

- **Japanese Total Access Communication System (J-TACS) and Narrow-Band Total Access Communication System (N-TACS)**

Japanese Total Access Communication System (J-TACS) and Narrow-Band Total Access Communication System (N-TACS) are single band frequency spectra currently used by Japan. The overall band occupies 10 MHz of spectrum. This allows 800 total channels. Channels are not separated into separate bands (A or B) in these systems. The original specification, J-TACS, uses only the even numbered channels. Only when the narrow specification N-TACS was implemented, did the odd channels come into use.

Channel spacing	25 kHz		
Duplex spacing	55 MHz		
Frequency range for voice and signalling channels:			
Base receive/mobile transmit	915.025-924.475 MHz		
Base transmit/mobile receive	860.025-869.975 MHz		
Voice channels: N-TACS	752	J-TACS	376
Signalling channels: N-TACS	48	J-TACS	24

J-TACS/N-TACS cell sites receive on the higher frequencies of the allocated spectrum transmit on the lower frequencies spectrum. Conversely, AMPS and TACS cell sites transmit on the higher frequencies and receive on the lower frequencies. As a result, J-TACS/N-TACS subscriber units cannot be used in TACS or AMPS systems. The same is also true of the subscriber units of the TACS and AMPS systems – they cannot be used in the J-TACS/ N-TACS systems.

- **Universal total access communication system (U-TACS)**

Universal Total Access Communication System (U-TACS) is used in Europe, some parts of Asia, the Middle East, China and Africa. U-TACS has a 15 MHz frequency spectrum using 25 kHz channel spacing, providing up to 920 channels. The U-TACS frequency spectrum and channel allocation is the same as the combined TACS/E-TACS with the exception of the lower 8 MHz of the spectra. With 8 MHz less spectrum, U-TACS has 320 fewer channels.

2.3.3.2 Second Generation Systems

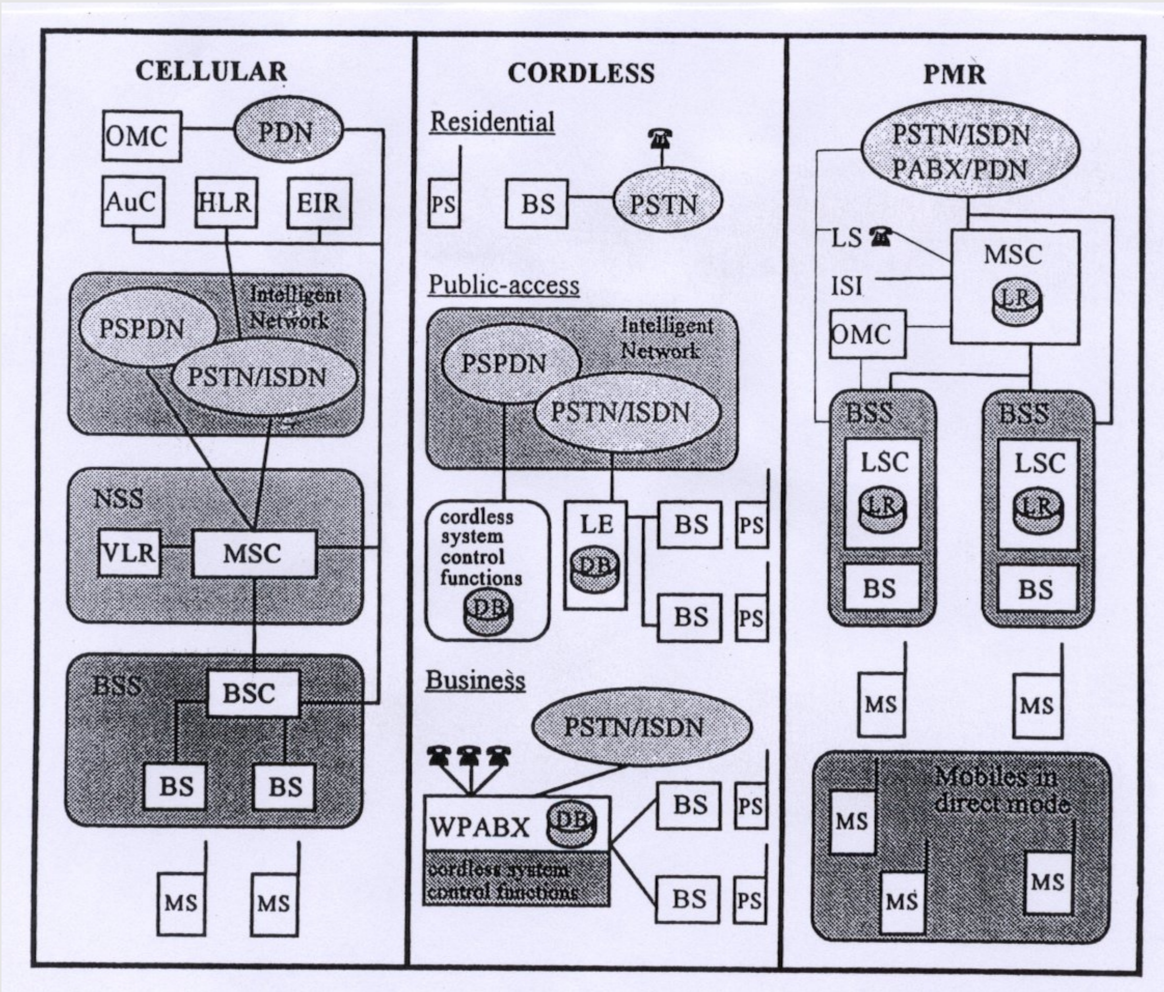
Designed during the 1980s, all of the so-called second generation mobile communication systems are digital. For voice calls, digitally encoded speech is transmitted on the radio interface using one of the many available digital modulation schemes. To offset the processing complexity required for these digital systems, two advantages are 1) the possibility of using spectrally efficient radio transmission schemes (e.g., time division multiple access [TDMA] or code division multiple access [CDMA]) in comparison to the analogue frequency division multiple access (FDMA) schemes previously employed; and 2) the ability to implement of a wide variety of (integrated) speech and data services and security features (e.g., encryption).

Typical cellular, cordless, and PMR network architectures are represented in Figure 2.3.3. These are reference models which may not always fully apply to all systems described hereafter.

In cellular systems, the base station subsystem (BSS) comprises a base station controller (BSC) and radio transceivers (BS or BTS) which provide radio communication with MS in the covered area. The network subsystem (NSS) includes dedicated mobile switching equipment (MSC) linking all system elements through leased lines to PSTN, ISDN, and Packet Switched Public Data Network (PSPDN). The home and visitor location registers (HLR/VLR) are databases containing mobile subscriber data and used for subscriber registration and mobility management. Copies of the subscribers' secret keys are stored in the authentication centre (AuC) and the mobile equipment serial numbers are stored in the equipment identity register (EIR). ITU-T Signalling System No. 7 (SS#7) and related application protocols are often used in the mobile network. All system elements are operated, controlled, and maintained by the operation and maintenance centre (OMC).

The cordless network architecture depends on application. For residential use, the portable station (PS) behaves like a regular telephone and has direct access to the PSTN through the private BS. In a public-access system, BSs are connected to a local exchange (LE) containing a local database (DB) used for subscriber registration and mobility management in the covered area. The LEs are connected to the PSTN/ISDN (for the purpose of traffic routing) and to centralized elements of the cordless system through the PSPDN (for signalling exchange). These centralized elements perform control functions (user identification, charging, network management) and may contain a centralized database that stores location updates of the cordless subscribers and, therefore, enables routing of incoming calls. For business applications, the same private automatic branch exchange (PABX) may be used for both wire and wireless access. The wireless PABX (WPABX) interconnects the BSs of the private network. Cordless subscribers can therefore access other private wired subscribers or the PSTN/ISDN. The WPABX generally incorporates the subscriber DB and control functions of the cordless system.

Figure 2.3.3 – Network reference models



The architecture of a PMR network is somewhat similar to that of cellular networks, but the BSS is generally consists of a single piece of equipment incorporating the BS and the local station controller (LSC). Since the LSC contains a copy of the subscribers location register (LR), local calls (which represent a significant part of the traffic) can be established and routed locally in the BSS, thus achieving a short call set-up time and maintaining local operation, even in fallback mode, if the BSS-MSC link is interrupted. Inter-site calls are routed through the MSC and access to other networks (PSTN/ISDN/PDN), or devices (PABX) may be provided either at MSC or BSS level. Line stations (LS) may be connected directly to BSS or MSC, or through an intervening network (e.g., ISDN). Interworking of different PMR networks of the same standard is provided through the intersystem interface (ISSI). Mobiles may also use direct simplex communications (direct mode), either autonomously or while keeping contact with the network (dual watch).

The major second generation systems are described in Table 2.3.2:

Table 2.3.2 – Air interface characteristics of second generation systems

	Cellular				Cordless			PMR	
Standard	GSM1800 (DCS)	IS-54	IS-95	PDC	CT2	DECT	PHPS	TETRA	APCO Project 25
Frequency band (MHz)	Europe,	USA	USA	Japan	Europe & Asia	Europe	Japan	Europe	USA
Uplink	890–915 (1710–1785)	824–849	824–849	940–956 (1429–1441, 1453–1465)	864–868	1880–1900	1895–1907	380–400?	Various bands, e.g. 150–170 ~800
Downlink	935–960 (1805–1880)	869–894	869–894	810–826 (1477–1489, 1501–1513)					
Duplex spacing (MHz)	45 (95)	45	45	130 (48)	—	—	—	10?	?
Carrier spacing (kHz)	200	30	1250	25	100	1728	300	25	12.5 (6.25)
No. of radio channels in the frequency band	124 (DCS: 374)	832	20	640	40	10	77	?	several hundreds of channel pairs
Multiple access	TDMA	TDMA	CDMA	TDMA	FDMA	TDMA	TDMA	TDMA	FDMA
Duplex mode	FDD	FDD	FDD	FDD	TDD	TDD	TDD	FDD	FDD
Number of channels per carrier	8 (half rate: 16)	3 (half rate: 6)	MABC	3 (half rate: 6)	1	12	4	4	1
Modulation	GMSK	Π/4 DQPSK	QPSK BPSK	Π/4 DQPSK	GFSK	GFSK	Π/4 DQPSK	Π/4 DQPSK	C4FM or CQPSK
Carrier bit rate (kb/s)	270.8	48.6	1288	42	72	1152	384	36	9.6
Speech coder Net bit rate kb/s)	RPE-LTP 13	VSELP 7.95	QCELP (var. rate: 8, 4, 2, 1)	VSELP 6.7	ADPCM 32	ADPCM 32	ADPCM 32	ACELP 4.5	IMBE 4.4
Channel coder for speech channels	1/2 rate convol. + CRC	1/2 rate convol. + CRC	1/2 (down) 1/3 (up) convol. + CRC	1/2 rate convol. + CRC	no	no	no	2/3 & 4/9 rates convol. + CRC	Golay & Hamming codes
Gross bit rate speech+channel coding (kb/s)	22.8	13	var. rate 19.2, 9.6, 4.8, 2.4	11.2	—	—	—	7.2	7.2
Frame size (ms)	4.6	40	20	20	2	10	5	57	20
MS transmission power(W)	Peak Aver. 20 2.5 8 1 5 0.625 2 0.25 DCS1800 1 0.125 0.25 0.031	Peak Aver. 9 3 4.8 1.6 1.8 1.6	0.6	Peak Aver. 2 0.66	Peak Aver. 0.01 0.005	Peak Aver. 0.25 0.01	Peak Aver. 0.08 0.01	Peak Aver. 10 2.5 3 0.75 1 0.25	?
Power control MS	Y	Y	Y	Y	low power mode in MS	N	Y	Y	N
control BS	Y	Y	Y	Y		N	Y	N	N
Operational C/I (dB)	9	16	6	17	20	12	26	19	?
Equalizer	needed	needed	Rake receiver	option	no	option	no	option	no
Handover	Y	Y	Soft handoff	Y	N	Y	Y	option	option

2.3.3.2.1 Digital Cellular Mobile Networks

2.3.3.2.1.1 Global System for Mobile Communications/Digital Cellular System 1800 (GSM/DCS1800)

The GSM standard was specified by the European Telecommunications Standards Institute (ETSI) for pan-European digital cellular mobile radio services. It was designed to answer the need for a common mobile communications standard throughout Europe, where a variety of incompatible analogue cellular systems like the Nordic Mobile Telephone (NMT) and the Total Access Communications Systems (TACS) existed. The allocation of a dedicated pan-European frequency band around 900 MHz and the signature of a Memorandum of Understanding (MoU) between countries, which committed to launch nation-wide GSM networks with internetworking capabilities, were the other important steps leading to an European network.

Phase 1 of the GSM standard was completed by ETSI in 1990 and is the basis of currently implemented networks. It provides a variety of speech and data services. These services, progressively being offered to the users, include telephony, emergency calls, conference calls, fax transmission, short messages, and data transmission at various rates up to 9600 bit/s. Supplementary services like call forwarding, call barring, and connected line identification are also defined.

The GSM network architecture closely follows the general principles introduced in the preceding section. All interfaces between network elements have been standardized, including MSC-BSC (A) and BSC-BTS (Abis) interfaces. Interfaces to AuC/HLR/EIR and to PSTN /ISDN use SS#7 with the mobile application part (MAP) protocol for non-circuit related signalling. This architecture, with a clear breakdown between the different equipment and different functional parts (e.g. radio resource management in the BSC), facilitates evolution.

The subscriber identity module (SIM) is the key to personal mobility, whereby a user can use any GSM terminal equipment just by inserting his SIM card. This smart card contains all subscriber data and is also used for basic security functions, such as subscriber identity authentication and key generation for traffic encryption on the air interface. This prevents fraudulent use of the system and ensures call privacy.

The GSM air interface is characterized by an eighth-order TDMA scheme with frequency division duplex (FDD). The available frequency band in Europe is 2×25 MHz, with a radio channel spacing of 200 kHz. Data is modulated at 270 kbit/s using Gaussian minimum shift keying (GMSK) modulation and transmitted in bursts of 577 µs. Each TDMA frame consists of eight time slots corresponding to eight separate physical channels. Each of these physical channels supports a combination of logical channels that are used in turn to carry signalling or traffic data. Slow frequency hopping is used to combat adverse propagation conditions, and most infrastructure implementations also include BTS receiver antenna diversity. The relatively low carrier to interference ratio (C/I) operational value (9 dB) is achieved through powerful channel coding, interleaving, and equalization techniques. Speech transmission is based on a linear prediction coder called regular pulse excited-long term prediction (RPE-LTP), which yields a net bit rate of 13 kbit/s and a gross bit rate of 22.8 kbit/s after channel coding. The air interface protocol follows a classical layered structure and includes a number of advanced features that are specific to mobile radio applications, such as mobile assisted handover (MAHO), power control (in both up and down links), and discontinuous transmission (DTX) based on voice activity detection (VAD).

GSM standardization in ETSI is still an ongoing process, and a number of additional services and features (such as multiparty calls, half-rate coder, and general packet data service) will be available in phase 2. Standardization of a GSM adaptation tailored to railway application has also recently started. This will include specific features such as group calls or support of high-speed mobiles.

One important GSM extension is the digital cellular system-1800 (DCS1800) standard designed for personal communication networks ([PCN], optimized for urban and suburban use) and for which various licences have already been granted in Europe. The main differences with GSM are the frequency band (around 1800 MHz), national roaming capabilities, and a reduced transmission power (hence a reduced cell size). It is also a candidate standard for the U.S. Personal Communication Services (PCS) in the 1900-MHz band.

Originally focused on the European market, the GSM/DCS1800 standard has now achieved worldwide recognition and deployment. More than 65 countries have already adopted the GSM standard, and at least 40 GSM/DCS1800 networks are currently in service around the world. These figures are constantly growing. (See Fascicle 2 – Chapter 3 for more details)

2.3.3.2.1.2 Interim Standard (IS-54)

The main driving force for the definition of a second generation digital standard in North America was the rapidly growing demand for cellular services during the 1980s. This would have easily exceeded the capacity of the analogue networks based on the advanced mobile phone system (AMPS). Accordingly, the new digital standard was specified by the Telecommunications Industry Association (TIA) upon request of the Cellular Telecommunications Industry Association (CTIA), with the major objective being to provide a significant increase in system capacity while maintaining upward compatibility with wide-spread AMPS. The Federal Communications Commission (FCC), however, decided to open the existing cellular band (2×25 MHz in the 800-MHz range) to any suitable technology.

The IS-54 standard was finally selected from several proposals and published in January 1991. Both dual mode (AMPS/IS-54) mobile stations and base stations are specified, thus enabling the design of equipment capable of analogue or digital operation. Other related standards have also been defined: IS-55 and IS-56 were defined for the performance specifications and measurement methods for mobile stations and base stations, respectively. Concerning network aspects, several standards have been developed by TIA since 1988 independently from the air interface design and are, hence, applicable to analogue AMPS as well as IS-54 and other systems.

Though less ISDN oriented than GSM, the IS-54 standard also supports several services, e.g., telephone service, short message service, and data services with a maximum transmission rate of 9.6 kbit/s. Supplementary services include call forwarding, three-party call, and call barring. Security features include a personal identification number (PIN), subscriber authentication upon connection to the system, and voice as well as subscriber data encryption.

The IS-54 air interface uses TDMA/FDD technology with three channels per 30-kHz AMPS carrier. The modulation bit rate is 48.6 kbit/s. $\pi/4$ shifted differential quadrature phase shift keying (DQPSK) modulation is employed. For each (full rate) channel, the gross bit rate is 13 kbit/s, and speech is encoded at 7.95 kbit/s using a vector sum excited linear prediction (VSELP) algorithm.

Advanced radio-link control based on power control and DTX improves spectrum efficiency. The air interface protocol, compatible with the AMPS protocol, includes an optional extended mode to allow for the addition of new system features and operational capabilities.

The traffic capacity achievable with IS-54 is expected to be three to four times that of existing AMPS systems. This capacity will be doubled with the introduction of a half-rate codec, currently under standardization. Using the original frequency band of the AMPS system, digital channels compliant to the IS-54 standard are now progressively replacing analogue channels, thereby alleviating the shortage of spectrum while enabling a smooth transition from analogue to digital. Network and terminal equipment are already available from several manufacturers and commercial service is being offered in the largest U.S. cities.

Based on the IS-54 standard, Hughes has developed a new technology called E-TDMA, with operational networks in the U.S. and recently adopted for several regional networks in Russia and China. It is claimed to achieve a significantly higher capacity, taking advantage of advanced features such as half-rate coding, digital speech interpolation (DSI) and channel pools.

2.3.3.2.1.3 Interim Standard IS-95

Introduction

Code division multiple access (CDMA) provides a spectrally efficient, digital solution for the second generation of cellular, wireless telephony, and personal communications systems (PCS) services. The CDMA air interface is near optimum in its use of the subscriber station transmitter power, enabling the widespread commercial use of low-cost, lightweight, hand-held portable units that have improved battery life. The technology is also near optimum in its link budgets, minimizing the number of base stations required for an excellent grade of service coverage. As customer penetration in a given service area increases, the system is called upon to deliver more capacity. CDMA has demonstrated a capacity increase over advanced mobile phone service (AMPS) of at least a factor of ten, which means that up to ten times fewer base stations will be required when customer demand for service increases. The use of soft handoff nearly eliminates the annoyance of dropped calls, fading, and poor voice quality.

CDMA, in this context, means not just generic code division multiple access, but the specific implementation of it described in the air interface standard: *TIA/ EIA/ IS-95-A: Mobile Station – Base Station Compatibility Standard for Dual-Mode Wideband Spread Spectrum Cellular System.* Compliance with the requirements in this standard assures subscribers and service providers that equipment of all types will interoperate satisfactorily.

IS-95-A originated with a system design developed by QUALCOMM, beginning in April 1989. The early requirements were defined by the cellular carriers with input from manufacturers and carriers and by public testing. After the initial proposal and presentation of a draft air interface, a standards committee composed of system operators, subscriber equipment vendors, infrastructure equipment vendors, and test equipment vendors reviewed, revised, and formalized the air interface. The formal adoption of what is now the IS-95-A air interface took place in December 1993.

The CDMA System

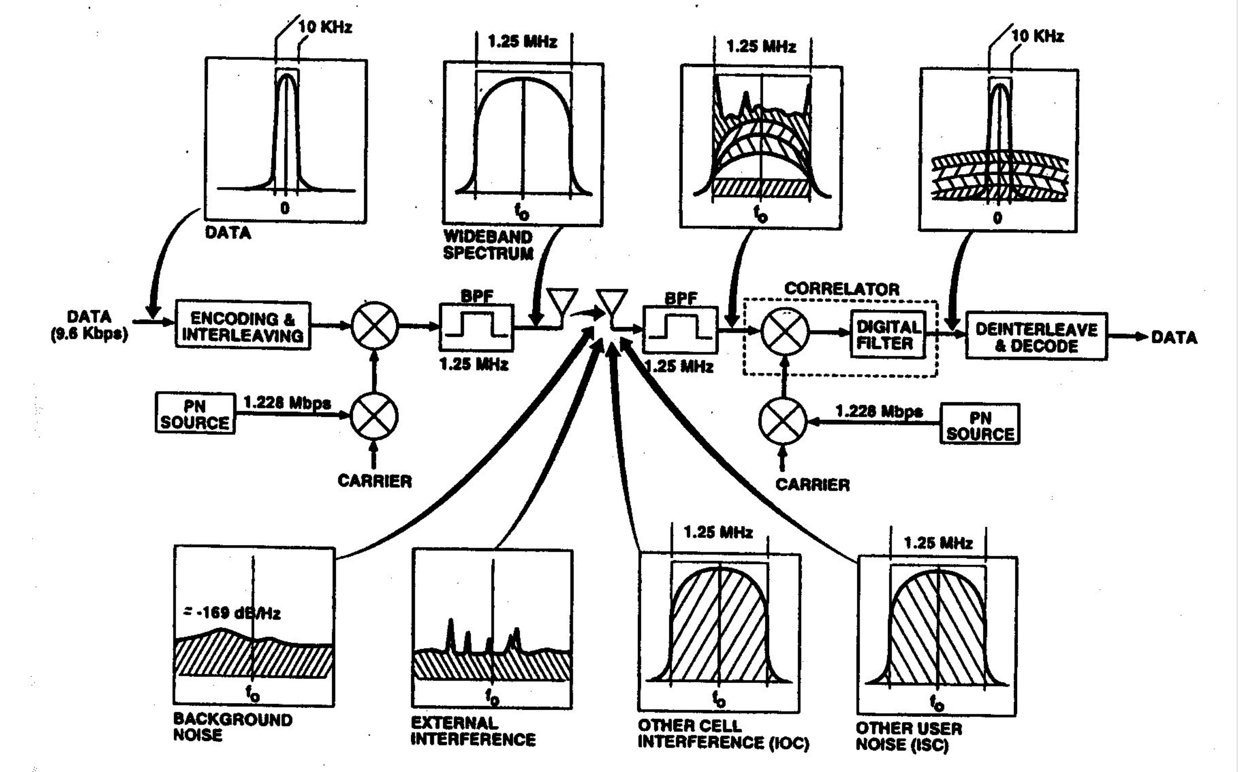
The multiple access scheme exploits isolation provided by the antenna system, geometric spacing, power gating of transmissions by voice activity, power control, an efficient modem, and a signal design that uses powerful error correction coding.

A combination of open-loop and closed-loop power control (through measurements of the received power at the mobile station and the base station) commands the mobile station to make power adjustments in order to maintain only the power level required for adequate performance. This minimizes interference to other users, helps to overcome fading, and conserves battery power in the mobile station.

The CDMA digital cellular waveform design uses a pseudo-random noise (PN) spread spectrum carrier. The chip rate of the PN spreading sequence was chosen so that the resulting bandwidth is about 1.25 MHz after filtering, or approximately one-tenth of the total bandwidth allocated to one cellular service carrier.

The USA Federal Communications Commission (FCC) has allocated a total of 25 MHz for mobile station to cell site and 25 MHz for cell site to mobile station, for the provision of cellular services. The FCC has divided this allocation equally between two service providers, the A and the B carriers, in each service area. Because of the time sequence of the FCC’s actions in allocating the cellular spectrum, the 12.5 MHz allocated to each carrier for each direction of the link is further subdivided into two sub-bands. For the B carriers, the sub-bands are 10 MHz and 2.5 MHz each. For the A carriers, the sub-bands are 11 MHz and 1.5 MHz each. A signal bandwidth of less than 1.5 MHz fits into any of the sub-bands, whereas a bandwidth of less than 2.5 MHz fits into all but one sub-band.

Figure 2.3.4 – View of CDMA concept



View of the CDMA concept. The desired signal is selected from four different sources of interference. The dominant source is system self-interference produced by other users of the same cell. This source is controlled by closed-loop power control.

A set of ten 1.25-MHz-bandwidth CDMA channels can be used by each operator if the entire allocation is converted to CDMA. Initially, only one or a small number of 1.25 MHz channels needs to be removed from the present PM analogue service to provide digital service. This facilitates the deployment by introducing a gradual reduction in analogue capacity. Each 1.25 MHz CDMA segment can provide about twice the capacity of the entire 12.5 MHz allocation using the present FM system. Some frequency guard band is necessary if there are adjacent high-power cellular (or other) frequencies in use, and the maximum capacity of the CDMA cell is required. Capacity can be sacrificed for decreased guard band if desired. Adjacent CDMA channels need not employ a guard band.

Overview of the IS-95 CDMA Standard

The common air interface standard, IS-95-A, prescribes, in detail, the behaviour of dual-mode CDMA/AMPS subscriber stations, and, to a lesser extent, the base stations. Use of compliant subscriber stations assures base station manufacturers of predictable behaviour on which to base system designs. An IS-95-A compliant subscriber station can obtain service by communicating with either an AMPS (analogue PM) base station or with a CDMA base station. System selection depends on the availability of either system in the geographic area of the station, as well as its programmed preference. Minimum performance requirements for dual-mode base stations, IS-97, and subscriber stations, IS-98, supplement the basic air interface.

IS-95-A emphasizes subscriber station requirements because the subscriber side reflects all call processing features, and because the specification is simpler in the context of a single user. In contrast to the detailed subscriber station requirements, base station requirements are incomplete. Generally the standard prescribes only those base station requirements that are important for the design of subscriber stations, leaving unspecified behaviour to the discretion of the vendors. Base stations are fielded in much smaller quantities, but at much greater cost per unit. Market considerations therefore encourage good designs.

The standard specifies that mobile stations operating with analogue base stations meet the analogue compatibility provisions for mobile stations as specified in EIA/ITA/IS-54-B, *Bual-Mode Mobile Station-Base Station Compatibility Specification,* January 1992. The incorporation of the analogue portions of EIA/ITA/IS-54-B instead of EIA/ITA-553 *(Mobile Station-Land Station Compatibility Specification,* September 1989) accommodates all the changes to analogue operation imposed by the EIA/ITA/IS-54-B dual-mode standard.

2.3.3.2.1.4 Personal Digital Cellular (PDC)

As in North America and in Europe, the design of the PDC standard in Japan was motivated in the late 1980s by the saturation of analogue cellular networks and by the need for new and enhanced services. After a study phase initiated by the Japanese Ministry of Posts and Telecommunications in April 1989, the PDC air interface standard was issued in April 1991 by the Research and Development Centre for Radio Systems (RCR) under the name STD27 [RCR-STD27, 1991]. It was complemented by network interface specifications providing the basis for a unified digital cellular system in Japan and enabling connectivity with fixed ISDN.

RCR STD27 is a common air interface specification. Though there are some similarities with the American IS-54 standard in terms of technical features, no compatibility with existing analogue systems was required here. The new digital systems in Japan benefit from a specific spectrum allocation, initially in the 800-MHz band and later in the 1.5-GHz band.

The carrier spacing is 25 kHz and the multiple access scheme is a TDMA/FDD of order 3 with the current full-rate codec and of order 6 with the future half-rate codec. The carrier bit rate is 42 kbit/s. $\pi/4$ DQPSK modulation is used. The full-rate speech codec employs a VSELP algorithm with a gross bit rate of 11.2 kbit/s and a net bit rate of 6.7 kbit/s, with forward error correction based on a convolutional code (rate ½) and a cyclic redundancy check (CRC). A specific channel assignment procedure (flexible channel reuse from one BS to another) enables an increase of the system capacity. The standard also specifies power control and MAHO type handover procedures.

PDC systems will provide numerous services, including speech transmission, data transmission (G3-facsimile, modem, videotex), and short message service. Supplementary services such as calling line identification, call forwarding, or three-party call are also foreseen. The air interface protocol is ISDN oriented with a layered structure following the open systems interconnection (OSI) principles, including a link access protocol called LAP DM at layer 2 and a layer 3 divided into radio transmission management (RT), mobility management (MM), and call control (CC, based on ITU-T I.451). Security features include authentication and encryption.

Interfaces between network elements of a PDC system have been defined by cellular operators in Japan, except the A interface (BSS-MSC) which is left open for implementation. The network architecture follows the reference model of ITU-T SS#7*,* if used between network elements and on the interface to other networks. The application protocols are an enhanced version of ISDN user part (ISUP) for circuit related signalling and MAP, developed as an application service element on TCAP, for non-circuit related signalling.

Commercial service with a PDC network was initiated by NTT in 1993 for the 800-MHz band and in 1994 for the 1.5-GHz band. Two other operators have launched digital cellular services in the 500 MHz band in 1994, and the Japanese government has recently decided to allow two new operators to offer digital service in the 1.5 GHz band. Enhancements of the PDC standard are also foreseen (such as the half-rate codec, packet data, etc.).

2.3.3.3 Cordless Telephony (CT)

The first generation of the UK's cordless telephones (coded CT1) was developed as the answer to the large quantities of imported, technically superior, yet unlicensed mobile radio equipment. The simplicity and cost effectiveness of CT1 analogue radio and base station products using eight RF channels and FDMA scheme stem from their applications limited to incoming calls from a limited number of mobile users to the isolated telepoints. As the number of users grew, so did the co-channel interference levels, while the quality of the service deteriorated. Anticipating this situation, the second generation digital cordless telecommunications radio equipment and *common air interface* standards (CT2/CAI), incompatible with the CTl equipment, have been developed. CT2 schemes employ digital voice but the same FDMA principles as the CTl schemes. Network and frequency re-use issues necessary to accommodate anticipated residential, business, and telepoint traffic growth have not been addressed adequately. Recognizing these limitations and anticipating the market requirements, different FDMA, TDMA, CDMA, and hybrid schemes aimed at cellular mobile and digital cordless telecommunications (DCT) services have been developed. The technical characteristics of some schemes are given in Table 2.3.3:

Table 2.3.3 – Comparison of Digital Cordless Telephone Systems

Comparison of Digital Cordless Telephone Systems

Parameter	CT2Plus	CT3	DECT	CDMA
Multiple access method	(F/T)DMA	TDMA	TDMA	CDMA
Duplexing method	TDD	TDD	TDD	FDD
RF channel bw, MHz	0.10	1.00	1.73	2 × 1.25
RF channel rate, kb/s	72	640	1152	1228.80
Number of traffic ch. per one RF channel	1	8	12	32
Burst/frame length, ms	1/2	1/16	1/10	n/a
Modulation type	GFSK	GMSK	GMSK	BPSK/QPSK
Coding	Cyclic, RS	CRC 16	CRC 16	Conv 1/2, 1/3
Transmit power, mW	≤ 10	≤ 80	≤ 100	≤ 10
Transmit power steps	2	1	1	many
TX power range, dB	16	0	0	≥ 80
Vocoder type	ADPCM	ADPCM	ADPCM	CELP
Vocoder rate, kb/s	fixed 32	fixed 32	fixed 32	up to 8
Max data rate, kb/s	32	ISDN 144	ISDN 144	9.6
Processing delay, ms	2	16	16	80
Reuse efficiency[3]				
Minimum	1/25	1/15	1/15	1/4
Average	1/15	1/07	1/07	2/3
Maximum	1/02[1]	1/02[1]	1/02[1]	3/4
Theor. number of vc. per cell and 10 MHz	100 × 1	10 × 8	6 × 12	4 × 32
Practical per 10 MHz				
Minimum	4	5–6	5–6	32 (08)[2]
Average	7	11–12	11–12	85 (21)
Maximum	50[1]	40[1]	40[1]	96 (24)

[1]The capacity (in the number of voice channels) for a single isolated cell.
[2]The capacity in parentheses may correspond to a 32 kb/s vocoder.
[3]Reuse efficiency and associate capacities reflect our own estimates.
Source: 4U Communications Research Inc., 1995.02.23–22:39 updated: 1994.10.31.

The complete description of these systems is given in the Chapter 4 of Fascicle 2.

2.3.3.4 Private Mobile Radiocommunications (PMR)

The world of PMR is manifold as there are so many different user groups with very different operational requirements. PMR is used by large organizations devoted to public safety, such as police, customs, fire brigades, rescue and ambulances etc. who need hierarchically structured networks with local, regional or even nation-wide coverage. There are also non-emergency authorities like government departments, public health, environment protection, PTTs and the like with similar networks. However, PMR is also used by medium and small user groups with very differing requirements for a wide range of applications.

PMR employs a variety of different technologies and covers a wide field of applications from local communication of small user groups up to networks with nation-wide coverage.

Definitions

To understand why PMR is versatile and unique it is necessary to understand the underlying operational and economical requirements and the resulting properties in detail. Firstly a *short definition* of PMR should be given:

PMR offers two-way radiocommunications carrying speech, data or a mix of both in non-public networks tailored to the specific operational requirements of professional mobile user groups for efficient and flexible communication within their area of daily operation.

Until around 1980 PMR systems were mostly systems employing a single base station or repeater, the modulation was analogue FM or PM, and the modes of operation were simplex, half or full duplex. Up to 80 mobile stations were served per radio channel and the coverage could reach up to 30 km. PMR meant mainly PTT controlled voice transmission, with a relatively limited amount of proprietary data services.

More recently paging and data facilities, digital switches and trunking, as well as PSTN and ISND connections were added. Sophisticated digital modulation including non-constant envelope schemes have been introduced and digital signal processing permits the use of transmission techniques giving more channels and less interference. Moreover, integrated data services like X.25, X.400 etc. are now used to a larger extent and closed user groups (CUG) with dynamic reconfiguration are now possible.

The use of a common channel is widespread and a decisive factor in the spectral efficiency of PMR. In many applications channel access must be available almost immediately, e.g. in less than 200 ms. It is important to note that PMR traffic is very different from cellular services, the talk time amounts to seconds instead of minutes with a much higher call frequency, hence the channel access time must be in a reasonable relation to the talk time. The coverage must be reliable and well suited to the users' needs including the requirements for coverage in areas with bad propagation conditions. Since PMR comprises a variety of radiocommunication systems tailored to the different needs of various user groups several main applications can be distinguished.

Self-provided large PMR systems – consist of networks having typically more than 300 mobiles in densely populated areas. Utilities, public urban transport, railways, power suppliers and large industrial sites are the main user groups. Specific requirements including limited access to PSTN and ISDN characterise the needs of such groups.

Small PMR networks – have less than 50 mobiles. Taxis, vehicle recovery services and business users are typical customers of such systems. The migration to GSM makes the future growth of this segment uncertain.

Service-provided PMR systems – are often called Public Access Mobile Radio *(PAMR) or CBS* (Community Base Stations). They are trunked systems run by an operator and have been in operation since the mid eighties. The technical and licensing conditions vary from country to country. Where PSTN or ISDN access is possible, these systems could be regarded as a public service competing with GSM. There are many PMR and PAMR networks with regional limited coverage, however, there are also networks operating nationwide for police and other emergency bodies.

Spectrum Allocation

PMR originally was allocated in the band 40 to 80 MHz as a *national* matter. Hence there was no harmonisation resulting in currently three different channel separations and many other differences. Later the PMR allocations were extended considerably.

Equipment

a) Analogue equipment

Conventional single channel PMR systems suffer traffic congestion and inefficient use of spectrum. Trunked systems use channels for control as well as traffic. The advantages are more efficient use of the spectrum and hence more traffic per channel. In order to save frequencies and thus be able to satisfy more users in the allocated PMR bands, the transition to 12.5 kHz channel separation instead of the 20 or 25 kHz was made during the seventies and the eighties in most European countries, and in most of the PMR bands. However in reality the overall gain in spectrum efficiency is about 1.1 instead of 2.0 if fading and shadowing are taken into account.

Currently the trunked PMR systems MPT 1327 and MPT 1343 employing analogue speech transmission and signalling at 1200 bit/s dominate in the PAMR sector and several different variations are available. Trunked PMR generally is dominated by proprietary protocols. Conventional PMR users are also moving towards trunked system solutions because of the better frequency utilisation.

b) Digital equipment

The first attempt to create a complete digital PMR *(DPMR) system standard* was made by ETSI with DSRR for 25 kHz channels in the 900 MHz employing GMSK and the GSM speech coder. TETRA (Trains European Trunk Radio System) is the second *DPMR system standard*, called now EP TETRA, for DPMR equipment supporting all kinds of digital transmission, e.g. data and digitized voice. The results of the standardization work on TETRA are in the ETSI standards series ETS 300 392 to 396.

i) *TETRA* is a TDMA system with four times slots per carrier operating on 25 kHz channels for 385-470 MHz, 870-921 MHz. The gross bit rate is 38 kbit/s offering usable bit rates up to 28.8 kbit/s with time slot aggregation. Single time slot traffic channels provide a gross bit rate of 9 kbit/s and an unprotected user bit rate of 7.2 kbit/s.

ii) Additionally two de facto PMR standards have been developed based on proprietary specifications, namely MPT 1327 and 1343 (initiated by the UK DTI and providing digital calling and data transmission at 1.2 kbit/s combined with analogue voice) and MOBITEX (which was developed originally only for data at 1.2 kbit/s, while MOBITEX II or MOBITEX 8k offers a gross bit rate of 8.0 kbit/s on 25 kHz channels in the 400 and 900 MHz bands).

iii) Additionally, several proprietary fully digital PMR systems have been launched during the last few years. The most important of the proprietary DPMR systems in the market place are listed below:

   – *APCO 25* is a fully digital FDMA system developed for the USA, for the 150, 400 and 800 MHz bands, and based on 12.5 kHz channel separation and C4FM modulation. A future migration to 6.25 kHz channels and a linear modulation scheme (CQPSK) is envisaged.

   – *ASTRO* is a FDMA system developed by Motorola for DPMR applications in the 2 m band and for the three channel separations 12.5, 20 and 25 kHz using QPSK-C modulation and providing a gross bit rate of 9.6 kbit/s.

   – *EDACS* is the Enhanced Digital Access Communications System developed by Ericsson with 12.5 kHz and 25 kHz channel separation in the 160, 450, 800 and 900 MHz bands. The system uses GFSK modulation and is upgradeable channel by channel from analogue FDMA also providing TDMA. The TDMA version was launched in 1995 and has become fully digital. EDACS is a multi-site system and offers a gross bit rate of 9.6 kbit/s and large area coverage.

– GeoNet is *a FHMA system* provided by Geotek. It is based on Frequency Hopping Multiple Access and 800 and 900 MHz band implementations are available. It is claimed that such a network with 50% data traffic exhibits a capacity increase by up to 30 times compared with trunked analogue PMR. (This is possible due to the much higher transmission rate compared to systems like MPT 1327).

– *MIRS* is the Motorola Integrated Radio Service which was introduced by Motorola in 1991 in the US. In 1995 MIRS was renamed iDEN (Integrated Dispatch Enhanced network). It is a 6TDMA linear modulation system employing m16QAM (m=4) modulation and providing a gross bit rate of 64 bit/s, and thus a six fold capacity increase compared with traditional PMR systems is claimed, being increased further by appropriate geographic frequency reuse to 18 times. This system has been developed for the 800, 900 and 1,500 MHz bands.

– TETRAPOL was developed by Matra and has been available since 1992. It is based on FDMA and available for the 80 MHz band and the 400 MHz range with a channel separation of 12.5 kHz. A channel separation of 10 kHz is also feasible on demand. The gross bit rate is 8.0 kbit/s based on GMSK modulation. Many of its current applications are devoted to public safety.

Modern digital PMR systems are structured according to the ISO/OSI layer model in a systematic and transparent order and offer a variety of specialised services. The *Basic Services* comprise *Tele Services* and *Bearer Services* while *the Supplementary Services* consist of *PMR Type Supplementary Service* and *Telephone Type Supplementary Services*.

Teleservices offer different encrypted or non-encrypted types of calls, mainly individual calls (point to point), group calls (point to multipoint), acknowledged group calls and broadcast calls, while the bearer services comprise circuit mode unprotected or protected data and packet mode connection oriented or connectionless data, usually available with different network transmission speeds, e.g. for TETRA 2.4 to 9.6 kbit/s in case of protected data and 7.2 to 28.8 kbit/s for unprotected data in circuit mode.

There are also a variety of PMR type supplementary services available. The most interesting ones are access priority, pre-emptive priority, priority call, call authorised by dispatcher, include call, transfer of group control, late entry, ambience listening, discreet listening, area selection, short number addressing, talking party identification and dynamic group number assignment.

Lastly a comprehensive list of telephone type supplementary services is provided, e.g. list search call, call forwarding, call barring, call report, call waiting, call hold, calling party identity presentation, calling party identity restriction, call completion, advice of charge and call retention.

This variety of offered services indicates the complexity of modern digital PMR systems. It also shows that such systems do not only offer all the usual PMR features but also some additional advanced services and a wide range of telephone services.

2.3.3.5 Third Generation Mobile Systems – IMT-2000

IMT-2000 third generation mobile systems will deliver voice, graphics, video and other broadband information direct to the user, regardless of location, network or terminal. These fully personal communication services will provide terminal and service mobility on fixed and mobile networks, taking advantage off the convergence of existing and future fixed and mobile networks and the potential synergies that can be derived from such convergence. The key benefits that IMT-2000 promises include improvements in quality and security, incorporating broadband and networked multimedia services, flexibility in service creation and ubiquitous service portability.

Networked multimedia may be defined to include services such as pay- TV; video- and audio-on-demand; interactive entertainment; educational and information services; and communication services such as video-telephony and fast, large file-transfer.

For more information on IMT-2000 see Fascicle 2 – Chapter 3, and the Final Acts of the World Radio-communications Conference 2000.

2.3.3.6 The Use of Satellite Systems

The systems employ one or more satellites to serve as base station(s) and/or repeater(s) in a mobile radio network. The position of satellites relative to the service area is of crucial importance for the coverage, service quality, price, and complexity of the overall network. When a satellite encompasses the Earth in 24-h periods, the term *geosynchronous orbit* has been used. An orbit that is inclined with the respect to the equatorial plane is called an inclined orbit; an orbit with a 90° inclination is called a *polar orbit.* A circular geosynchronous 24-h orbit over the equatorial plane (0° inclination) is known as *geostationary orbit,* since from any point at the surface of the Earth the satellite appears to be stationary; this orbit is particularly suitable for the land mobile services at low latitudes and for maritime and aeronautical services at latitudes of < 80°. Systems that use geostationary satellites include INMARSAT, MSAT, and AUSSAT. An elliptical geosynchronous orbit with the inclination angle of 63.4° is known as *tundra orbit.* An elliptical 12-h orbit with the inclination angle of 63.4° is known as *Molniya orbit.* Both tundra and Molniya orbits have been selected for the coverage of the northern latitudes and the area around the North Pole; for users at those latitudes, the satellites appear to wander around the zenith for a prolonged period of time. The coverage of a particular region *(regional coverage)* and the whole globe *(global coverage)* can be provided by different constellations of satellites including those in inclined and polar orbits. For example, inclined circular orbit constellations have been proposed for GPS (18-24 satellites, 55-63° inclination), Globalstar (48 satellites, 47° inclination), all systems will provide global coverage. ORBCOM system employs Pegasus launchable low-orbit satellites to provide uninterrupted coverage of the Earth below ± 60° latitudes and an intermittent but frequent coverage over the polar regions.

Satellite antenna systems can have one *(single-beam global system)* or more beams *(multibeam spot system).* The multibeam satellite systems similar to the terrestrial cellular system, employs antenna directivity to achieve better frequency reuse, at the expense of system complexity.

For more details, see Subchapter 2.4 of Fascicle 1 and the Final Acts of the World Radiocommunications Conference 2000.

2.3.4 Abbreviations

ADPCM	Adaptive Differential Pulse Code Modulation
AMPS	Advanced Mobile Phone System
ANSI	American National Standards Institute
ASIC	Application Specific Integrated Circuit
AuC	Authentication Centre
BSC	Base Station Controller
BSS	Base Station Sub-system
CAD	Computer Aided Design
CBS	Community Base Station
CC	Call Control
CDMA	Code Division Multiple Access
CELP	Code Excited Linear Prediction
CRC	Cyclic Redundancy Check
CT	Cordless Telephony
CTIA	Cellular Telecommunications Industry Association
CUG	Closed User Group
DAMA	Demand Assigned Multiple Access
DB	Database

DCS	Digital Cellular System
DCT	Digital Cordless Telecommunications
DPMR	Digital Private Mobile Radio
DQPSK	Differential Quadrature Phase Shift Keying
DSI	Digital Speech Interpolation
DSP	Digital Signal Processing
DTX	Discontinuous Transmission
EIR	Equipment Identity Register
ETACS	Expanded Total Access Communication System
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FHMA	Frequency Hopping Multiple Access
FSK	Frequency Shift Keying
FM	Frequency Modulation
GSM	Group System Mobile
HLR	Home Location Register
IMT	International Mobile Telecommunications
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
JTACS	Japanese Total Access Communication System
LAP	Link Access Protocol
LE	Local Exchange
LR	Location Register
LSC	Local Station Controller
MAHO	Mobile Assisted Handover
MAP	Mobile Application Part
MIRS	Motorola Integrated Radio Service
MMIC	Microwave Monolithic Integrated Circuit
MoU	Memorandum of Understanding
MPEG	Moving Picture Expert Group (ISO)
MSC	Mobile Switching Centre
NMS	Network Management System
NMT	Nordic Mobile Telephone
NSS	Network Sub-system
NTACS	Narrow-band Total Access Communication System
OSI	Open Systems Interconnection

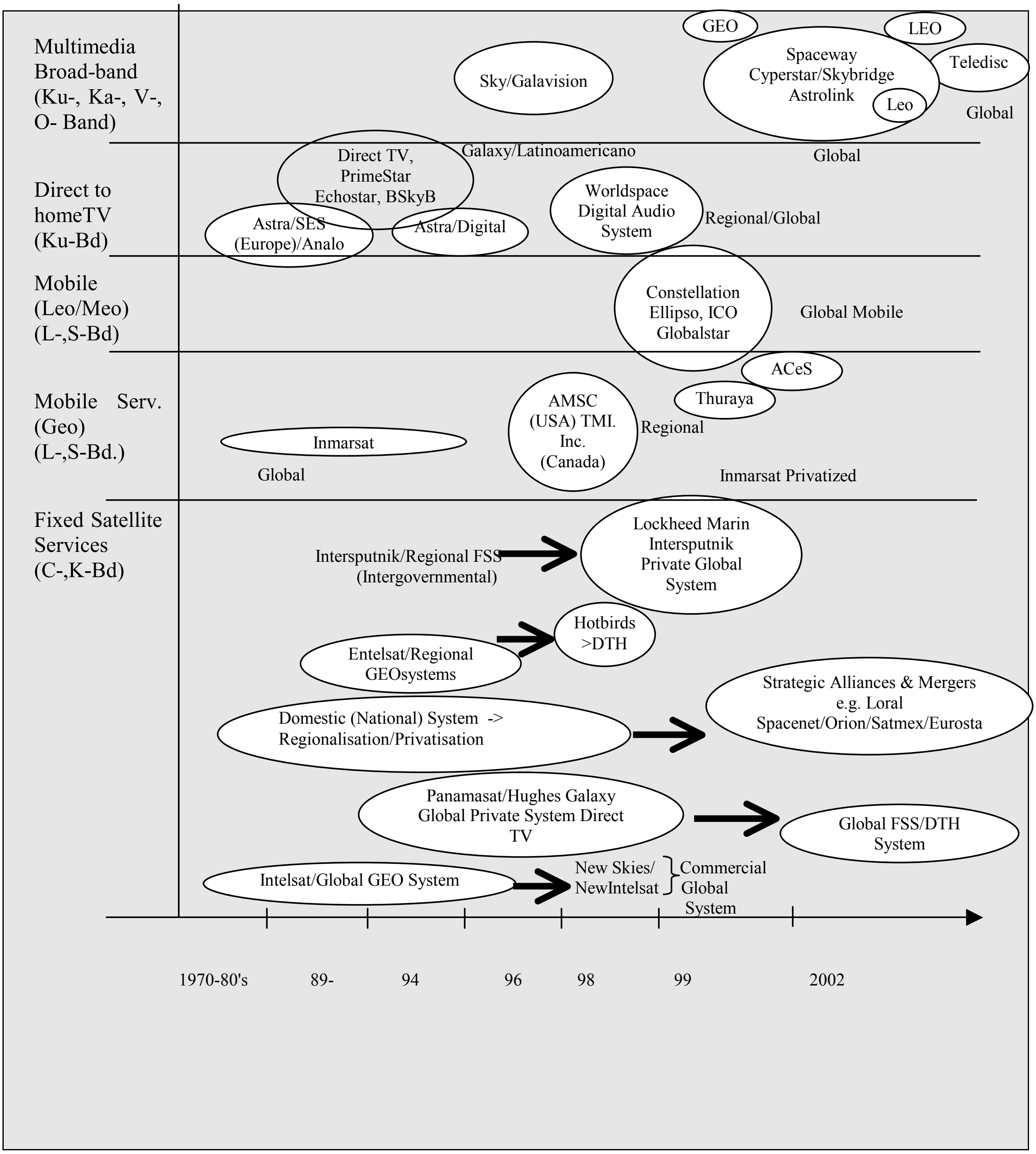
PAMR	Public Access Mobile Radio
PCS	Personal Communication System
PDC	Personal Digital Cellular
OMC	Operation and Maintenance Centre
PABX	Private Automatic Branch Exchange
PM	Phase Modulation
PMR	Private Mobile Radio
PS	Portable Station
PSTN	Public Switched Telephone Network
PSK	Phase Shift Keying
PSPDN	Packet Switched Public Data Network
QPSK	Quadrature Phase Shift Keying
QAM	Quadrature Amplitude Modulation
RDR	Research and Development Centre for Radio Systems
RPE-LTP	Regular Pulse Excited – Long Tern Prediction
RT	Radio Transmission
SS7	Signalling System No. 7
TACS	Total Access Communication System
TCAP	Transaction Capability Application Part
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TETRA	Trans-European Trunk Radio System
TIA	Telecommunications Industry Association
VAD	Voice Activity Detector
VLR	Visitor Location Register
VSELP	Vector Sum Excited Linear Prediction
WPABX	Wireless Private Automatic Branch Exchange

2.4 Satellite systems

Global communications coverage by satellites has been available for many years using large earth stations. This is now being extended to smaller and smaller mobile terminals. The three most suitable categories of satellite orbits for communications purposes are:

- Geostationary Orbit (GSO)
- Non-Geostationary Orbit (Non-GSO)
  1) Medium Earth Orbit (MEO)
  2) Low Earth Orbit (LEO)

Table 2.4.1 – Development of Satellite Systems



Satellite Orbits

- If the GSO is used, the entire world except for the polar regions can be covered by only 3 satellites in equatorial orbits. The altitude of these satellites must be approximately 35,800 km above the surface of the earth. The satellites will appear to an observer on earth as being stationary.

- If a MEO constellation is used, the altitude has to be selected between the inner and outer Van Allen Radiation belts, typically around 10,355 km above the surface of the earth. The orbit period in this case will be 6 hours. The world can be covered by 10-12 satellites in 2-3 planes e.g. 5 satellites in each of 2 planes or 4 satellites in each of 3 planes.

- If a LEO constellation is used, the satellite altitude is much lower, typically between 700 to 2000 km. The orbit period will be somewhere between 100 and 120 minutes. Due to the lower altitude, the view of the world from the satellite is rather small and the number of planes to cover the entire globe will have to be increased to 6-8, and the number of satellites must be 6 per plane.

Typical representations of satellite system technology development as well as future trends are shown in Figure 2.4.1.

In the sections below the technology of GSO and non-GSO satellite systems are explained.

2.4.1 GSO Satellites

The beginning of satellite communications was dominated by the GSO satellite concept. The Syncom-2 satellite launch on July 19, 1963 was the first geosynchronous satellite (33 degrees orbit inclination), whereas Syncom-3, launched on August 19, 1964 was the first geostationary satellite.

The main advantages of GSO satellites are the following:

- Three operational satellites are sufficient to provide full-coverage of the earth, excluding the polar caps.

- GSO satellites are fixed in the sky, so it is not necessary to provide large earth station antennas with fast moving auto-tracking systems, as would be the case with LEO satellites.

- The earth station antenna is required to work with only one GSO satellite for a continuous connection, whereas with LEO satellites it is either necessary to use an earth station antenna able to jump quickly from the setting satellite to a rising one, or to use two antennas for each earth station.

- The onboard antenna of the GSO satellite may be very directive. A LEO satellite, due to its moderate altitude and quick movement with respect to earth, is forced to use a small-gain antenna or fast moving tracking systems if the onboard antenna gain is increased.

GSO satellites have some disadvantages with respect to LEO satellites:

- The propagation delay is much larger and may cause an echo with about 500-ms time deference;

- The free space attenuation is much larger. The distance between a point on the earth and the satellite will vary between a minimum of 35,786 km for the equatorial point aligned with the satellite and the centre of the earth, and a maximum of 41,756 km for the points located on the cone projected from the satellite and tangent to the earth surface. The free space attenuation for a GSO satellite link varies therefore for the various frequency ranges;

- They do not cover the polar caps;

- They may not provide coverage of big cities for land-mobile communications, due to the shadows created by tall buildings.

2.4.1.1 Fixed GSO Satellite Systems

Non-mobile satellite systems will continue to depend on satellites in GSO orbits, but as the lower microwave bands (e.g. C-Band) allocated to such systems increasingly become more crowded, fixed satellite systems will be expected to move up to higher frequency bands (e.g. Ku-band). Therefore, attention has centred on developing equipment for new and higher microwave bands or increasing the capacity of existing frequency bands. Two typical and different communications systems which use advanced digital concepts are:

a) Demand Assignment Multiple Access (DAMA); and

b) Time Division Multiple Access (TDMA) system.

In the DAMA concept, a pool of satellite transponder channels is assigned to a number of users. DAMA provides instant, dial-up connectivity between a larger community of users and is an attractive option for any user looking for cost-effective digital technology combined with the lowest equipment investment. Operating costs are also low since DAMA circuits are charged according to the amount of satellite resources actually used, on a per-minute basis.

TDMA is a digital service for public switched networks which require high connectivity on medium density traffic routes. DAMA is an attractive option for users who want the highest quality INTELSAT service because of higher stability, greater equipment flexibility and better digital bit error ratio (BER) than any other INTELSAT service. The quality and availability of this service is consistent with current fibre optic technology.

The search for more capacity in a given bandwidth is shown in the following examples:

Antenna technology

More capacity is made available by designing antennas for higher microwave bands, and by reusing frequency bands with different polarizations and spatial beam separations. This technology is equally applicable to satellite and earth station designs.

Low noise and high power amplifier technology

As satellite transponders become more powerful and earth stations more sensitive, the number of communications channels that can be squeezed into a given bandwidth increases.

High precision frequency oscillators and converters

Digital communications systems are highly dependent on the stability and purity of oscillators and clock sources derived from the oscillators. For example, in high-capacity TDMA (Time Division Multiple Access) systems as used in satellite systems, the arrival time at the satellite transponder of transmitted bursts from numerous participating earth stations is controlled to within microseconds despite physical movements of the satellite, Doppler shifts and other propagation abnormalities. If this strict control was not possible, more guard-time between bursts would have to be allocated with an ensuing loss of channel carrying capacity.

Modem technology

An important development in modem technology is to reduce its C/N (Carrier-to-Noise) threshold for a certain Bit Error Rate (BER). If the modem is operating at a low C/N ratio, it will need less power from the satellite, which means that the power can be used for additional satellite channels. Advancements in modem technology are also closely related to FEC (Forward Error Correction) technology.

Use of Digital Signal Processing

FEC technology is just one example of technology which can be implemented using digital signal processing.

Other advancements in this field are the use of digital signal processing for the compressing and decompressing of signals to ever lower transmission rates. This technology is applied to any type of analogue signals such as voice-band channels or TV signals, and since lower transmission rates mean smaller transponder bandwidth the number of channels in a given bandwidth may be increased.

An additional example of digital signal processing used in fixed satellite systems is in Digital Circuit Multiplication Equipment (DCME) introduced in this section. By using this technology, the number of telephone or data channels carried by a satellite RF channel may be increased by a typical factor of 5.

2.4.1.2 Digital Circuit Multiplexing Equipment

One of the major advantages of converting communications systems from analogue to digital technology is that it opens up a wide range of possibilities in signal processing. Three processing concepts which, when taken individually, have already made important improvements in digital communications, have had a significant impact on digital system designs when they are combined. These three processing concepts are (1) Low Rate Encoding (LRE) and compression/decompression techniques, (2) Digital Speech Interpolation (DSI), and (3) Forward Error Correction (FEC) techniques. The equipment which exploits the first two processes when combined is known as Digital Circuit Multiplication Equipment (DCME).

FEC is incorporated in the RF digital carriers transmitted over the satellite link.

The two main technologies combined in the DCME equipment, as mentioned above, are LRE and DSI. LRE refers to the reduction in the number of bits required to transmit a given piece of information. The standard 64 kbit/s PCM channel is produced by sampling an analogue input channel 8000 times per second and converting each sample to an 8-bit PCM code. However, by using a variety of techniques, it is possible to significantly reduce the number of transmitted bits for each sample. The reduction is dependent on the required grade of service and whether speech, voice-data or signalling will be transmitted. A reduction from 64 kbit/s down to 8 kbit/s is possible, and the ITU-T is developing a speech coder operating at 4 kbit/s.

DSI uses dynamic assignment techniques, i.e. a time slot (satellite channel)is assigned to a particular international communication channel on demand, but the same time slot is not used for the duration of the telephone call. This technique is made possible by the fact that during a telephone conversation, the speech is not continuous and so the channel is not required all the time and therefore the time slot can be allocated to a different channel. The DSI circuits concentrate a number of terrestrial input channels (generally known as trunk channels) into a smaller number of time slots (referred to as bearer channels or satellite channels). In the opposite transmission direction, the concentrated bearer channels are expanded into a larger number of trunk channels. The maximum number of satellite channels (time slots) is 61 (81 under overload condition).

TSI (Time Slot Interchange) in the DCME equipment is used as a static assignment technique, i.e. terrestrial channels are manually assigned to international channels until a manual reassignment is required. The terrestrial input lines to the DCME equipment are coming in 2 Mbit/s (typically up to 10) carriers, which may contain many empty time slots. Therefore, the potential number of terrestrial channels (typically 310) is higher than the 216 international channels that can be handled by the DSI circuits. The static assignment process (TSI process) is the manual procedure of picking out 216 international channels from the maximum 310 terrestrial channels. Hence, by static assignment techniques (TSI), 216 international channels are picked out from among 310 terrestrial channels, and by dynamic assignment techniques (DSI), the 216 international channels are compressed into 61 (or 81) satellite channels.

A certain trunk channel is connected to a certain bearer channel only for the time duration (including an appropriate hang-over time period) that the trunk channel is active, i.e. while it is carrying a burst of speech or transmitting voice-band data or signalling. Taking account of the interval between speech bursts during a normal telephone conversation, the trunk channel will be active only for about 30-40% of the time. For a large number of trunk channels, therefore, the number of bearer (satellite) channels can be reduced significantly (by a factor of 2.5-3.0) by assigning the satellite bearers “on demand”. Pre-assignment of channels should, however, be possible in the equipment.

Frequent trunk/bearer assignment information must be transmitted between the corresponding earth stations in order to make sure that the transmitting trunk channel is connected to the correct receiving trunk channel at the remote side for the duration of the telephone conversation, or voice-band data transmission. The information is exchanged in a special Assignment Channel (AC).

On the terrestrial interface side, each DCME module is connected to an International Switching Centre (ISC) with up to ten 2 Mbit/s communications links (total of 310 voice or data channels (64 kbit/s) plus 20 synchronization and signalling channels). On the satellite interface side, each DCME module is connected to an Intermediate Data Rate (IDR) modem operating at 2.048 Mbit/s information data rate.

DCME equipment may be used in other than satellite communications systems (for example submarine cables). If used for satellite applications, the operational modes are the following:

– Single Destination (SD) mode;

– Multi-Clique (MC) mode;

– Multi-Destination (MD) mode;

– Mixed SD/MD mode.

The SD mode, as the name implies, is the simplest mode and is also the most economical for large users. The minimum requirement as far as DCME/IDR equipment is concerned, is one redundant set of DCME modules, and one redundant set of IDR modems. In addition, the required earth station equipment is up-converters and down-converters, low-noise amplifiers, power amplifiers and antenna. IDR/DCME equipment may be implemented in a new earth station design or incorporated into an existing station. In the latter case, station facilities such as terrestrial microwave link, power plant, and air conditioning may already be available.

In Multi-clique operation, the satellite bearer channels on the transmit side are divided into one or two separate blocks, or “cliques", each clique associated with a different destination. Since two destinations are served, the earth station will need two IDR demodulators. The two demodulators will provide the wanted as well as the unwanted channels at its output. In order to avoid sending the unwanted channels to the ISC and thereby reducing the capacity of the terrestrial link, only the wanted channels are picked out and sent to the ISC by a Digital Cross-connect or Digital Branch equipment. By placing the DCME equipment at the ISC rather than the earth station, the reduction in the number of required satellite channels is also extended to the terrestrial link.

Multi-destination operation may be used for up to 4 destinations per DCME set and is economical for small users with multi-destination requirements. All satellite bearer channels are subject to DSI processing regardless of the channel destination. Figure 2.4.2. shows a system configuration with 4 destinations. In this example, one IDR modulator and 4 demodulators are required for one DCME equipment set. The DCME may be located at either the earth station or at the ISC.

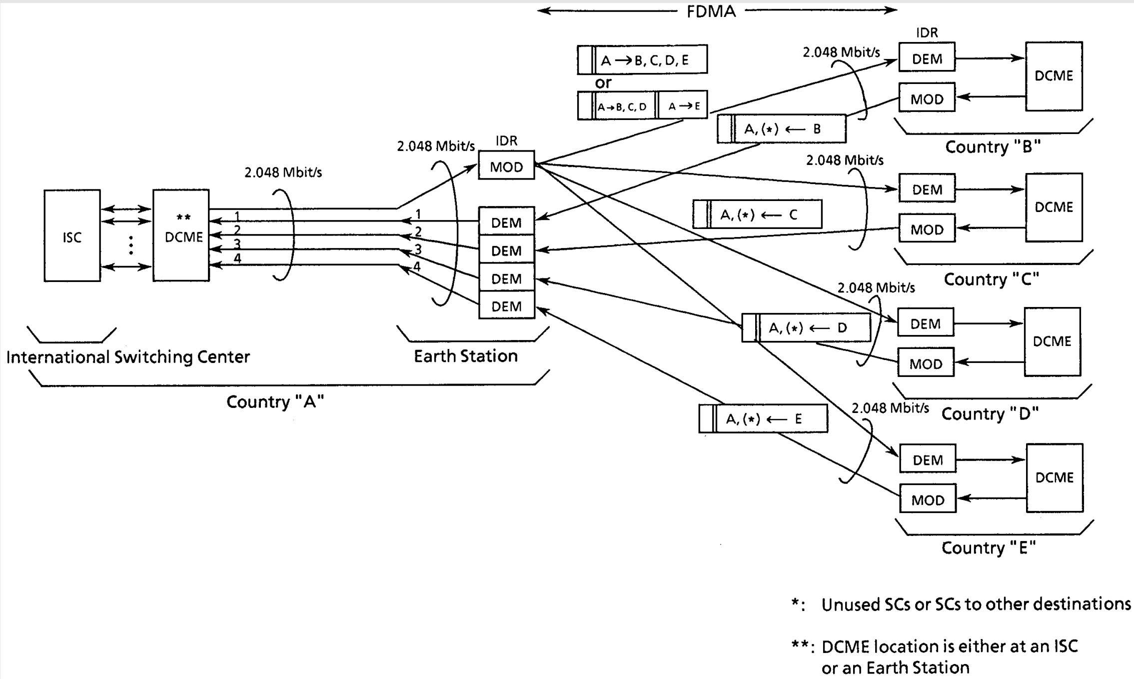
DCME in use on a satellite link should ensure provision for ISDN (Integrated Services Digital Network) bearer services. This means that the DCME must be transparent to circuit mode unrestricted 64 kbit/s terrestrial channels used with different access protocols. Likewise, the 64 kbit/s channel must be able to transmit 3.1 kHz voice-band data from modems. Furthermore, the DCME must handle alternate transmission of voice and data during the same call.

2.4.1.3 High-capacity International Satellite Telephony and Data Systems

The difference between satellite communications systems used for international links and for domestic or regional communications is primarily one of capacity. International earth stations are usually designed as high-capacity installations capable of handling from several hundreds to perhaps thousands of voice and data channels plus multiple TV channels. Domestic or regional systems, on the other hand are usually implemented for low to medium capacity perhaps from a few dozens to several hundreds channels, and a single TV channel.

Whether an earth station is designed for high capacity international links or for low capacity domestic or regional communications, the technology used in the earth station equipment is basically the same. The difference is in sizing the equipment to the application on hand plus employing equipment that will increase the number of channels on high capacity links.

Figure 2.4.2 – Basic concept of DCME/IDR multi-destination operation



This section contains information on the various equipment subsystems that are included in a typical earth station. The differences that characterize the large-capacity earth station from its low-capacity counterpart are the following:

- The physical size of the antenna is larger in order to increase the antenna gain and therefore the station's capacity. A typical antenna size is a diameter of 16-18 meters. The figure-of-quality of the station is indicated in its G/T figure, which is the receive antenna gain divided by the system thermal noise temperature in Kelvin. For example, a 16-meter antenna operating at 4 GHz may have a receive gain of 3.16×10^{-5} (55 dB) and a noise temperature at its operating elevation angle of 100 K. This noise temperature has two major components, namely the Low Noise Amplifier (LNA) temperature (for example 50 K), and the antenna noise temperature (50 K). This would give the station a G/T of 35 dB/K which is typical of an international earth station. The antenna feed and tracking equipment is identical to the ones described in the next section of this Handbook.
- The High Power Amplifiers (HPA) need a higher rated output power. Since large international stations often will be transmitting 2 or more RF carriers through the same HPA, intermodulation considerations may dictate a rather large HPA power back-off and the HPA must be sized accordingly.
- The modulator(s) and demodulator(s) must be able to handle high capacity information data streams of typically 2 048, 6 312, 8 448, 32 064, 34 368 and 44 736 Mbit/s. Even higher information rates such as 139.264 Mbit/s in the Plesiochronous Digital Hierarchy (PDH) and 155.52 Mbit/s in the Synchronous Digital Hierarchy (SDH) may find roles in satellite communications systems.
- High-capacity earth stations are often implemented with Digital Circuit Multiplication Equipment (DCME) which takes advantage of channel Low Rate Encoding (LRE) and Digital Speech Interpolation (DSI) technologies in order to increase the number of voice channels or voice-band data channels in a given satellite transponder bandwidth by a factor of up to 5.0.

High Capacity Modems

Modulator and demodulator (modem) technology has made great advancements during the last few years. Development has focused on reducing the physical size of the equipment and integrating numerous functions into Large Scale Integration (LSI) circuits. Such functions include Forward Error Correction (FEC) technology that makes it possible to operate the demodulator at ever decreasing Carrier-to-Noise (C/N) or Energy-per-bit-to-noise density (Eb/No) ratios for a certain Bit Error Rate (BER).

Modern high-capacity modems for earth stations are now available in a variety of configurations such as physically separate modulators and demodulators, combined modems, fixed transmission rate modems, variable rate modems, rack-mounted or plug-in card modems.

In the combined modem configuration, one modem in each shelf is stand-by for the remaining 8 on-line modems with automatic and manual switch-over control equipment built into the shelf. Likewise, IF combiners, dividers and a patch-panel are included in the shelf. In terms of channel capacity, then, one 210-cm rack may hold up to 32 on-line modems with a total maximum capacity of 3840 voice and/or data 64 kbit/s channels with 64 synchronization and signalling channels. In terms of capacity measured in bits per second, the total capacity of the rack is approximately 270 Mbit/s.

Whether the modem is implemented as a separate unit or combined, it is capable of operating with FEC rates of 1/2, 3/4 and 7/8 and may be implemented with concatenated Reed-Solomon coding technology. User interfaces are RS-422/449 or V.35 (48-8448 kbit/s) or ITU-T Rec. G.703 (1544-8448 kbit/s). For FEC rate 1/2 the demodulator will typically produce a BER of 1×10^{-6} or better at an Eb/No of 6.1 dB. The corresponding Eb/No figures for FEC R3/4 and FEC R7/8 are 7.6 dB and 8.7 dB, respectively.

The modulator is using 4-phase PSK (Phase Shift Keying) and the demodulator coherent 4-phase PSK technologies. IF frequencies are either 70 or 140 MHz, and the modem is capable of tuning in 2.5, 22.5 or 25 kHz steps across the IF passband. Frequency stability is typically $\pm 1 \times 10^{-6}$ per year. Furthermore, the modulator provides synchronization and ESC (Engineering Service Channel) signals for its corresponding demodulators, and drop and insert multiplexed signals are available in data rates between 64 and 1920 kbit/s. To keep demodulators in synchronization with modulators, Plesiochronous/Doppler buffers may be included. The power consumption of a fully equipped shelf (9 modem units) is approximately 0.8 KVA.

Extensive local and remote monitoring and control facilities of high-capacity modems are essential for their proper operation. Table 2.4.1 and 2.4.2 show typical items, which may be monitored and controlled in modern modem designs.

2.4.1.4 Low to Medium-capacity Regional and Domestic Satellite (DOMSAT) Systems

With the liberalization of telecommunications policies that has swept the world in recent years and with the prospects of even more liberalization to come, new satellite systems are being implemented around the world by organizations that previously, under monopolistic policies, would have been prevented from setting up their own systems. These systems, which normally have higher traffic capacity than VSAT (Very Small Aperture Terminal) systems can provide, are generally implemented within the borders of a certain country (Domsats) but may also cover a region of countries.

With the proliferation of new satellite systems and migration to higher frequency bands, there is a requirement for a more flexibility in earth station system design, and for the development of affordable yet ultra-reliable earth station subsystems since many stations will be operated unattended. Power consumption is also a significant aspect of new equipment design, because more and more earth stations will be using alternate primary power sources, such as solar power plants.

Table 2.4.1 – High-capacity modulator monitoring and control items

ITEM	CONTROL	MONITOR
Equipment status		0
Carrier frequency (Ch no.)	0	0
Transmit carrier level	0	0
Information data rate	0	0
FEC rate	0	0
Transmit framing type	0	0
Transmit terrestrial interface	0	0
Carrier on/off	0	0
Loop-back facility on/off	0	0
Scrambler on/off	0	0
AIS alarm		0
Data input loss		0
Backward alarm		0

Table 2.4.2 – High-capacity demodulator monitoring and control items

ITEM	CONTROL	MONITOR
Equipment status		0
Carrier frequency (Ch no.)	0	0
Information data rate	0	0
FEC rate	0	0
Receive framing type	0	0
Receive terrestrial interface	0	0
Loop-back facility on/off	0	0
De-scrambler on/off	0	0
Buffer reset	0	0
Bit error rate		0
Synchronization loss		0
AIS alarm		0
High bit error alarm		0
Backward alarm	4 destinations	0

The broad categories of equipment that comprise a modern digital earth station are the following:

1) Antenna system with feed and tracking equipment (if applicable),

2) Low-noise devices,

3) High-power devices,

4) Frequency up- and down-converters,

5) Modems (treated in the previous section of this book),

6) Centralized monitoring and control systems.

Flexibility in antenna technology can now be achieved by an antenna design that operates simultaneously in the C-band and the Ku bands, i.e. 6/4 and 14/11 GHz bands. This is important because many modern satellites will be equipped with both FSS frequency band transponders. Furthermore, the flexibility is advantageous to TV uplink and cable restoration earth stations that occasionally have a need for accessing different satellites operating with either C- or Ku-band satellites, or both at the same time.

Development of low-noise devices as the front-end of the earth station is focused on lowering the noise temperature, reducing the physical size, extending the Mean Time Between Failures (MTBF), and making the devices maintenance-free and capable of operating without cooling. MTBF figures of 500,000 hours for modern LNA (Low Noise Amplifiers) are not uncommon.

High-power amplifiers have developed from klystrons and travelling wave devices to FET transistor amplifiers under the generic term Solid State Power Amplifiers (SSPA). With this development, the physical size of the amplifiers has shrunk significantly, and the reliability has been extended enormously, primarily due to simpler power supply designs.

Frequency converters are essential components in an earth station for converting between common IF frequency bands and the assigned RF frequencies in the satellite transponder. With the change of communications technology from analogue to digital, the performance of the converters has taken on new significance with respect to phase noise, linearity, spurious output signals and reliability. Modern frequency converters make extensive use of advanced technologies such as monolithic FET RF amplifiers, monolithic bipolar IF amplifiers, and thin-film and thick-film Microwave Hybrid Integrated Circuits (MHIC).

Modulators and demodulator designs are important in digital earth stations because they, to a large extent, determine the overall quality of a satellite link, and they have the capability to lower the amount of RF power needed from the satellite, and maintain the communications link at a high quality level. This is accomplished by sophisticated Forward Error Correction (FEC) schemes that, in effect, lower the threshold of the demodulator to very small values.

Finally, centralized monitoring and control of one or several earth stations or a complete Domsat or regional system is important because the equipment is frequently unattended, not only in the main (central) earth station but very often also at remote stations under the control of the central station. A modern monitoring and control subsystem is computerized equipment (typically one or more PCs operating in a Local Area Network) from where a single operator can monitor and control all equipment in the main and remote earth stations. The software used for this purpose closely follows the development of modern-type windows software architectures with user-friendly man-machine interface.

Combined C/Ku-band Antenna Structures

To cater for different earth station system designs and different frequency bands, combined C- and Ku-band parabolic antenna technology with simultaneous dual-band, dual-polarization transmission and reception feeds is now available in antenna diameter sizes from 7.6 meters to 16 meters. Traditionally, dual circular polarization has been used at the 4/6 GHz C-bands, and linear dual polarization at the 11-12/14 GHz Ku-bands. With the increasing demands on the frequency bands, it is anticipated that linear and circular polarizations will also be used in the C- and Ku-bands, respectively. Likewise, it is anticipated that feed technology with switching between circular and linear polarizations will play a role in satellite systems.

Combined C/Ku-band Feed

The combined C/Ku-band feed is where most new antenna technological development has been invested. Figure 2.4.3 shows the basic concept of a combined dual-band/dual polarization parabolic antenna design. The feed is one integrated mechanical structure approximately 3 meters in length and containing only passive microwave devices.

Figure 2.4.3 – Combined C/Ku-band dual frequency/dual polarization antenna feed

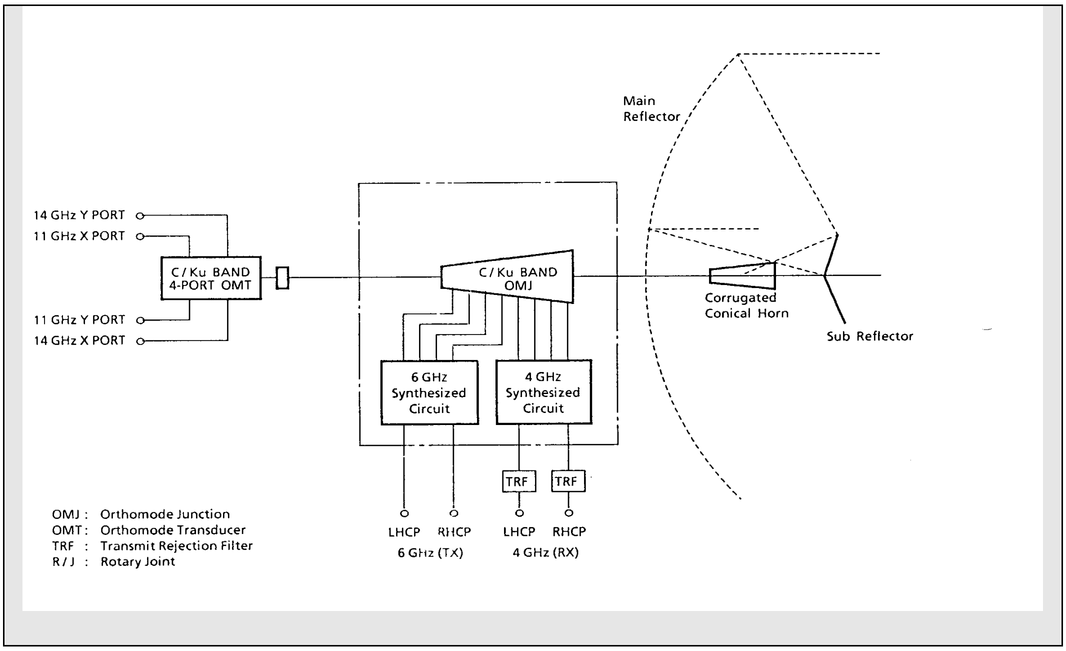


Table 2.4.3 gives the C- and Ku-band overall antenna transmit and receive gain figures in dB, and the thermal noise temperature in degrees Kelvin (K).

Satellite Tracking

Depending on the antenna diameter size of a given earth station and the movements of the applicable spacecraft, automatic/ manual satellite tracking may or may not be required, and the degree of sophistication in the tracking equipment may also vary from user to user.

At one end of the scale, small earth station terminals with antenna diameters of perhaps up to 5-6 meters and operating with highly geostationary satellites need no tracking equipment because their antenna beamwidth is so broad that the satellite never moves outside the beam. Or as a minimum tracking requirement, infrequent manual hand-crank or motorized antenna reorientations may be performed.

At the other end of the scale, large earth station installation of the Intelsat Standard-A class with antenna diameters of 16 meters or more, need automatic tracking equipment no matter what satellite will be used. This is due to the very narrow beamwidth of such stations. For example, a 16-meter Ku-band antenna will have a half-power beamwidth in the order of 0.12 degrees at 11.2 GHz. Simple manual and automatic tracking capability is a minimum requirement for such stations if they operate with stable geostationary satellites, but the degree of sophisticated technology in tracking equipment increases rapidly if a given earth station is to be operated with less stable or non-geostationary satellites; if the station is required to switch rapidly between two or more satellites; if it is located in a geographical area that often suffers signal attenuation and/or scintillations; or even if it is to be operated with satellites in orbits which are stable but inclined to the geostationary orbit.

Table 2.4.3 – Combined band overall antenna gain and thermal noise temperature

	Combined C/Ku-band Antenna Diameter Size																			
	7.6-meter				9.2-meter				11-meter				13-meter				16-meter			
	Gain (dB)				Gain (dB)				Gain (dB)				Gain (dB)				Gain (dB)			
4.0 GHz receive band	48.0				49.7				51.4				53.0				54.9			
6.0 GHz transmit band	51.7				53.7				55.4				56.9				58.1			
11.2 GHz receive band	55.2				57.6				59.6				61.1				61.2			
14.25 GHz transmit band	56.8				59.0				60.9				62.3				62.3			
Elevation (degrees)	5	10	20	40	5	10	20	40	5	10	20	40	5	10	20	40	5	10	20	40
4 GHz thermal noise (K)	59	46	40	37	58	45	39	36	57	44	38	35	56	44	38	35	58	46	41	38
11.2 GHz thermal noise (K)	93	75	63	56	90	73	61	54	89	72	60	53	89	72	60	53	94	77	66	59

Satellite tracking technology has evolved from analogue monopulse concepts based on early radar technology, to modern digital computer controlled step-tracking technology which is less expensive and technologically superior in the number of functions that can be implemented. Such functions are normally performed by a microprocessor-controlled Antenna Control Unit (ACU).

As a minimum, the ACU may be implemented for manual motorized tracking only, in which case a beacon down converter and a beacon receiver is not required unless a beacon signal level indication is desired. If automatic tracking is implemented, the ACU will judge the beacon DC level at regular time intervals. If and when the signal level drops to a predetermined level, the ACU will initiate step-movements of the antenna structure in azimuth and elevation until the signal level has been optimized. Such antenna movements are controlled from a Motor Control Unit which will apply power to antenna AC or DC drive motors. When the signal level is optimized, the ACU then will shut down the tracking equipment and wait for a predetermined time duration or until the signal level again is dropping.

Typically, one or more of the following tracking modes, depending on the applicable tracking requirements, may be implemented in a modern earth station:

- Manual Mode. From the front panel of the ACU, the earth station operator will manually move the antenna in azimuth and elevation to the desired antenna orientation.
- Automatic Mode. The tracking system is operating completely automatic and will follow any movement of the satellite.
- Preset Mode. The earth station operator will input from the front panel of the ACU a satellite position in azimuth and elevation. At the push of a button or at a predetermined time, the ACU will move the antenna to the preset position.

- Program Mode. In this mode, the earth station operator will input 11 ephemeral data parameters provided by the satellite operator. The ACU uses these data to calculate and point the antenna to the satellite position at regular intervals.
- Memory Mode. If this tracking mode is selected, auto-tracking must have been done for a minimum of the past 26 hours. The ACU will use this data to track the satellite from then on. This is done on the assumption that no, or very minor, orbit variations will occur from one day to the next.
- Intelligent Self-Learning Program Tracking Mode. This mode (which could appropriately be called an active tracking mode) is a further development from the memory tracking mode (passive). In this intelligent mode, the ACU will analyse the tracking data for the previous 24 hours and based on the result will predict the satellite position from then on. This mode is more accurate than any other mode including automatic tracking. Furthermore, it reduces wear and tear on the antenna mechanical components because the equipment is not activated as often. It may be operated without a beacon signal, and is particularly useful for earth station sites that suffer frequent scintillation problems.

The sophistication of the ACU and tracking system as a whole may be appreciated when it is considered that it has no way of knowing whether a DC signal drop is because:

1) the satellite is moving out of the antenna main beam,
2) the earth station antenna main beam is being reoriented, for example due to strong winds,
3) the beacon signal is suffering propagation abnormalities in amplitude and phase (particularly applicable to the 14/11 GHz systems),
4) the beacon signal is absent or faulty. The ACU must accommodate and solve all of these problems.

Low Noise Amplifiers/Converters

Successful satellite communications links rely on a tiny low noise Gallium Arsenide (GaAs) Field Effect Transistor (FET) installed at the front-end of an amplifier chain mounted in the antenna structure in close proximity to the feed. The amplifier chain is referred to as the Low Noise Amplifier (LNA). In some cases the received signals from the satellite are frequency down-converted in the same physical unit, in which case the front-end amplifier is referred to as a Low Noise Converter (LNC).

Low noise amplifier technology for satellite communications has evolved from narrow-band diode amplifiers physically cooled by liquid nitrogen or helium, to modern broad-band GaAs FET devices that require no cooling at all. At the same time, the physical size of the LNA has shrunk so that the amplifier practically is being built into the waveguide, in some designs.

At the Ku-band, earth stations may be implemented with either Ku-band LNAs of LNCs. If LNAs are selected, one amplifier unit is typically of the same physical dimensions as a C-band LNA. Such an LNA unit may have a noise temperature in the range of 80K through 180K and an overall gain of 55 dB. Depending on the application, the bandwidth of the LNA may be extremely wide, for example to cover all FSS frequency bands (10.95-12.75 GHz).

If the Ku-band earth station is implemented with a Low Noise Converter (LNC) as its front-end amplifier, the incoming 11-12 GHz signals from the satellite are down-converted in the LNC to a fixed IF band usually located around 1.0-1.5 GHz. Noise temperatures and gain are typically in the same order as the LNAs, i.e. 80K through 180K and gain of 55 dB depending on the application.

High Power Amplifiers (HPAs)

For earth stations that require FSS frequency band transmit power in the kilowatts, the High Power Amplifier (HPA) mainstay is the Travelling Wave Tube (TWT) or Klystron amplifiers. The design to choose for a particular application is dictated by technical characteristics, investment and operating costs.

With the advance of solid state technology, HPAs are increasingly being implemented in earth stations with Field Effect Transistor (FET) amplifiers which are slowly migrating from the lower microwave frequency bands to the Ku and higher bands with increasing power outputs. These amplifiers are generally assigned the generic name SSPAs (Solid State Power Amplifiers), and their significance is due to good technical performance, low power consumption and high reliability. For the same reasons, they are also increasingly being used as power amplifiers in modern satellite transponders.

A current state-of-the-art High Power Amplifier implemented in a new earth station should include the following features:

- High reliability and MTBF (Mean Time Between Failures)
- High gain and high gain stability
- Low intermodulation distortion
- Low power consumption
- Low AM/PM conversion
- Low residual AM and phase noise
- Low input surge current
- Low spurious output signals
- Simple operation and maintenance procedures
- Adequate fault and status indicators
- Adequate protection and alarm circuitry with logic sequencing to prevent mis-operation
- Auto-recycling for automatic restart after power outages or other reasons
- Compact and integrated packaging
- Ability to be integrated into a redundant transmit chain with automatic switch-over between on-line and stand-by HPAs
- Ability to be monitored and controlled remotely.

Depending on which satellite is to be used and at what frequency band, the proper choice of HPA for a particular earth station can be made. Table 2.4.4 shows typical HPA output power capabilities at the frequency bands allocated to satellite communications and for the 3 main types of amplifier devices.

Frequency Converters

With the transformation of satellite communications from analogue to digital technology, the earth station's frequency converters have taken on increased importance.

Frequency converters include up-converters and down-converters for conversion between the RF bands and an IF band. The purpose of the up-converter is to place one or several RF carriers within the frequency passband of the satellite transponder assigned in the transmit direction. Similarly, the purpose of the down-converter is to receive one or several RF carriers within the frequency passband of a transponder assigned for the receive direction. As such, the up-converters and down-converters are normally tuned to the transponder's centre frequency. The IF is fixed at either 70 or 140 MHz. If operation with a standard 36 MHz satellite transponder is anticipated, an IF of 70 MHz will suffice. If it is desired to operate with wider frequency band transponders, like 72 MHz, an IF of 140 MHz is more appropriate.

Table 2.4.4 – HPA output power for various frequency bands

	Frequency Band	Output Power
TWT HPAs	5.850 - 6.425 GHz	700W, 3KW
	12.75 - 13.25 GHz 13.75 - 14.50 GHz	130W, 300W, 600W
	14.00 - 14.50 GHz	1KW
	17.20 - 18.10 GHz	400W
	30 GHz band	100W, 200W, 500W
KLYSTRON HPAs	5.850 - 6.475 GHz	1.7KW, 3.4KW
	14.00 - 14.50 GHz	2KW, 3KW
	17.30 - 18.10 GHz	1.5KW
	30 GHz band	350W, 450W
SSPA HPAs	1.626 - 1.661 GHz	5W, 50W
	5.850 - 6.425 GHz	10W, 20W, 50W, 100W
	14.00 - 14.50 GHz	6W, 15W, 35W

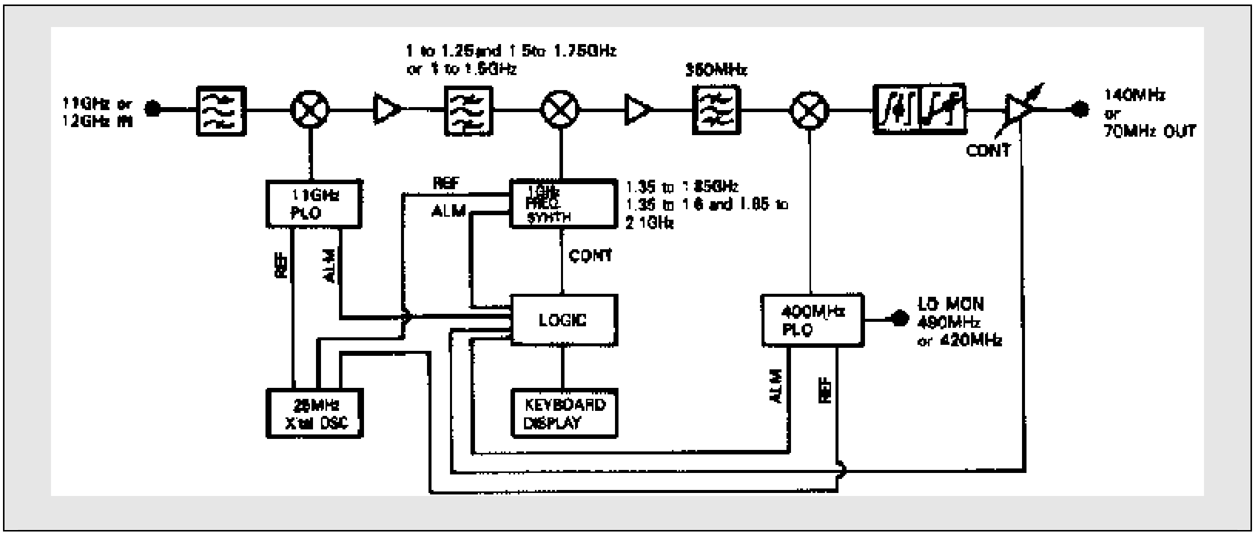
Modern up-converters and down-converters should include the following features:

- High reliability and MTBF (Mean Time Between Failures)
- High gain stability
- High linearity
- Low intermodulation distortion
- Flat IF/RF amplitude/frequency response
- Adequate IF/RF group delay response
- Low power consumption
- Low phase noise
- Low thermal noise
- Low spurious output signals
- Simple operation and maintenance procedures
- Adequate fault and status indicators
- Ability to be monitored and controlled remotely
- Compact and integrated packaging
- Ability to be integrated into a redundant transmit or receive chain with automatic switch-over between on-line and stand-by units.

Furthermore, the down-converter should be able to incorporate Automatic Gain Control (AGC) and Automatic Frequency Control (AFC) if not included in the basic design. Figure 2.4.4 shows a typical 12 GHz triple-conversion down-converter.

High reliability and MTBF figures can be assured by using thin-film and thick-film Microwave Hybrid Integrated Circuits (MHIC) as well as monolithic circuits such as GaAs FET and bipolar transistor amplifiers. For ease and flexibility of operation, a synthesized local oscillator should be used rather than a fixed pre-assigned local oscillator. Furthermore, a triple conversion design is advantageous for reducing the frequency steps in which the up-converter or down-converter can be tuned across its designed passband.

Figure 2.4.4 – Block diagram of Ku-band triple conversion down-converter



Modulators and Demodulators

Digital modulators and demodulators for domestic and regional satellite systems do not differ much in technology from their counterparts used in high-capacity systems as described in another section of this Handbook. The difference is rather in the amount of traffic they can carry.

A small-capacity domestic satellite system may be implemented with only a few pre-assigned digital voice channels, which may operate with an information rate of only 16 kbit/s. A higher capacity system may be designed for several hundred voice and data channels. Since it is not likely that all channels will be active at the same time and are not required between same geographical locations every time they are used, each channel may not be allocated permanent frequency space in the satellite transponder. Rather, the satellite channels will be assigned on demand, i.e. the satellite system will have a centralized *Demand Assignment Multiple Access* (DAMA) sub-system which will assign the channels as they are required. This will save satellite transponder bandwidth, and therefore operating costs for the operator of the satellite system.

Television programme distribution is a vital part of many domestic satellite systems, and with the currently available powerful satellite transponders, programmes can be provided to small roof-top TV receivers. Analogue TV transmission has been used since the beginning of satellite communications and is still being implemented in new earth station installations primarily due to the large amount of analogue TV receivers in operation in most countries. However, analogue TV will slowly be supplanted with digital TV which, when used with modern compression techniques such as MPEG-2, makes more efficient use of satellite bandwidth and power. Modulators and demodulators for digital TV will have data rates according to the image and sound quality desired. For example, a data rate of 64 kbit/s might be acceptable for a private PC-to-PC video conference, whereas a system that has to deliver studio-quality image and sound material would have to operate at a data rate of 2 Mbit/s or higher.

Centralized Monitor and Control System

In line with modern developments in operating and maintaining complex communications systems, sophisticated earth station Monitoring and Control (M&C) systems are often implemented such that one or more earth stations may be operated and maintained from a central location. This may or may not be at the location of the earth station(s). Such systems are also referred to as Computerized Station Management Systems (CSMS). M&C functions may be further extended to include monitoring and control of remote earth stations, which are under the control of one central earth station in a domestic or regional satellite system.

A modern CSMS system for monitoring and controlling an earth station (or complex of several earth stations) should include the following features:

- It should be tailor-made to the application on hand. The complexity of a modern earth station is such that a standard M&C system may not function very well.
- It should be comprehensive and cover the operation and maintenance of all equipment subsystems, whether the equipment is produced by one manufacturer, or includes equipment from several vendors. It should also be able to monitor auxiliary equipment such as fire and smoke detectors, and intruder detectors.
- It should have a high degree of flexibility. If the earth station is being reconfigured due to expansion or replacement of equipment, the CSMS should be able to be reconfigured as well, by earth station maintenance personnel, and without reprogramming of the CSMS software. A training function would also be a very useful feature.
- It should have extensive logging functions so that the performance of the earth station(s) can be analysed off-line.
- It should be easy to learn to operate. This means that it should use standard hardware to the extent possible, and it should use a standard computer user-friendly operating system which is in line with modern software developments.

2.4.1.5 VSAT Systems

A satellite VSAT (Very Small Aperture Terminal) system is an attractive solution for voice and data communications needs when the requirement is to have limited capacity from each location, but where the number of locations may run into the thousands. The flexibility of VSAT systems is such that it may be implemented very rapidly for a few sites and then gradually expanded as traffic grows or economic conditions allow. Therefore, this type of communications media is ideally suited for rural communications, when each small community may have requirements for only a few voice and data channels, but where there may be hundreds or perhaps even thousands of such communities scattered across the country.

The VSAT solution is also attractive when, in addition to voice and data communications, there is a need for one-way TV and radio broadcasting for educational or other purposes. In such cases, the remote VSAT terminals will serve not only as transmit and receive stations for voice and data, but also as TV and radio broadcast receive-only stations.

Furthermore, the digital data channels in modern VSAT terminals may be configured on-demand to serve specific purposes. This opens up a whole new spectrum of applications of which teleconferencing in general and telemedicine in particular are perhaps some of the most important. For example, a semi-skilled medical team with a video camera and medical equipment suitable for data communications, might visit VSAT communities on a regular basis. During the visit, two-way satellite video and voice channels will be made available for consultations between patients and skilled medical staff at central locations in the country.

The flexibility of VSAT systems extends to accommodate various types of traffic requirements. For example, some organizations (private or public) may have requirements for (1) voice communications and continuous data communications channels with various transmission rates, among many locations; whereas other organizations may have a need for (2) rapid and frequent interactive data communication, and occasional batch data transmission at fixed transmission rates between one central computer and many remote locations; or, (3) other organizations may have immediate requirements for both types of communications. Under traffic requirements, the routing of the traffic should also be examined. The VSAT system should be able to handle (4) *star* traffic in which all or most traffic flows between one central hub station and the remote VSAT terminals, and/or (5) *mesh* traffic, in which all or most traffic is evenly distributed among the VSAT terminals and the hub station. Also, (6) *broadcast* mode should be supported, in which the hub or control station is able to transmit to all or a predetermined group of VSAT terminals simultaneously. Finally, (7) it should be possible to implement digital or analogue television signal transmission by using the same RF equipment at the hub station and at the VSAT terminals, as used for the voice and data channels.

A well designed VSAT system should be able to implement all communications functions as outlined above. It should be able to handle various transmission rates between its users. Furthermore, the VSAT system should be able to use the available satellite channel (transponder or fraction of a transponder) in an economical way with respect to *power, bandwidth and time.* Finally, the VSAT system must be well *managed, monitored and controlled.*

Satellite transponder *power* is conserved by implementing the VSAT system with efficient Forward Error Correction (FEC) coding schemes which will allow the system's demodulators to operate at a very low Carrier-to-Noise (C/N) ratio and therefore need reduced transponder power. Furthermore, if the system is implemented for voice communications, the voice coding should be done at the lowest rate commensurate with the desired voice quality.

Since it is preferred that the VSAT system can accommodate various transmission rates for data as well as voice channels between its users, the satellite transponder *bandwidth* should be assigned on demand. This Bandwidth-on-Demand (BOD) concept should be extended to transmission rates between any modulator and demodulator in the system.

The satellite transponder's *time* should be used as efficiently as possible. This may be accomplished by implementing the VSAT system to be operated under DAMA (Demand Assignment Multiple Access) control, whereby satellite circuits (voice or data) are assigned only on demand (i.e. when needed).

For the VSAT system to function reliably, smoothly and with the minimum number of skilled personnel, it should have a built-in central Network Management System (NMS) from where the system can be *managed, monitored and controlled.*

VSAT systems may be implemented in the satellite communications allocated C-band as well as Ku-bands. Significant technological advancement has gone into the development of the hardware and software used in VSAT systems. Systems may be implemented as new stand-alone projects, or they be integrated into already existing earth station equipment, thus saving capital and implementation time.

A detailed description of one particular implementation of a VSAT System is given in Annex A.

2.4.1.6 Transportable DSNG TV Earth Station Transmission System

To meet the growing demand for on-the-spot TV coverage, equipment with video and sound compression techniques according to MPEG-2 (more specifically ISO/IEC 13818-2) standards is now available in a very compact form. To keep the dimensions and weight down to manageable levels and to take advantage of modern day high powered satellites, the equipment operates in the Ku-bands allocated to satellite communications.

The equipment can be broadly categorized into the Outdoor Unit (ODU), and the Indoor Unit (IDU).

The Outdoor Unit

The ODU contains the antenna which, depending on the available satellite capability, may be a 75 or 120 cm diameter design typically operating in the 14.0-14.5 GHz transmit band, and the 12.25-12.75 GHz receive band. The station's front-end low noise transistor amplifier/converter subsystem has a nominal noise temperature of 120K, and the station will have a G/T of approximately 15.25 dB/K at 40 degrees elevation, and using the 75 cm antenna.

The transmit equipment includes a high power TWT amplifier/converter which, depending on the satellite capability, has a nominal output power of 50 or 100 watts. The maximum transmitted power from the station (saturated EIRP) will be approximately 54 dBW by using the 75 cm antenna and the 50-watt amplifier.

Satellite tracking is done manually, and the station is able to move freely 360 degrees in azimuth, and between 25 and 55 degrees in elevation. It will operate in wind up to 20 meters/second, and the weight of the ODU is approximately 50 kg.

The Indoor Unit

The IDU equipment is also housed in a weatherproof container of weight 60 kg and dimensions 52.2W × 49.4H × 50.2D centimetres. It is powered by 115 or 200 VAC and consumes approximately 1 kVA (including the ODU equipment). It accepts standard composite NTSC or PAL video and audio signals of levels 1 Vp-p and 0 dBm, respectively. These signals are compressed to one of four video coding rates, selectable by the operator of the equipment.

The video coding rate selected is primarily dependent on the available satellite power and bandwidth. Table 2.4.5 shows a table of the parameters used by the 4 different transmission modes. What is shown as the "Normal" mode, corresponds roughly to the video and sound quality of a typical household receiver. In this mode, the video coding rate is 7 Mbit/s and the transmission rate over the satellite is 11.3 Mbit/s. By using a convolutional Forward Error Correction (FEC) rate of 3/4, the required satellite transponder bandwidth is 8 MHz. Thus, 3 additional and similar TV carriers can be accommodated in a standard 36 MHz transponder, provided it has enough power. The demodulator will produce an output data stream with a BER (Bit Error Rate) of 1×10^{-8} or better at a received Carrier-to-Noise ratio (C/N) of 6.8 dB.

In transmission systems where the power from the satellite transponder is severely limited, the operator can select the "Threshold Mode", in which case the video coding rate is reduced and the transmission rate is increased compared with the "Normal" mode. The increase is due to the use of FEC rate 1/2, and the demodulator is able to operate at a C/N of only 3.6 dB for the same BER as for the "Normal" mode.

If enough satellite transponder power and bandwidth is available, the operator may select the "Medium" or "High Quality" modes of transmission which use the transmission parameters as indicated in the table. The "Medium Quality" mode is roughly equivalent to the image quality produced by today's full transponder analogue FM TV transmissions. The "High Quality" mode will produce images and sounds which, although not comparable to High Definition TV (HDTV), will satisfy most users.

Audio is provided at a bit rate of 384 kbit/s, either as 2 monaural channels or as 1 stereo channel. Orderwire communications are included with four 32 kbit/s voice channels and one 64 kbit/s data channel.

Mobile Transportable Earth Station

The equipment as described above may be modified and installed in a road vehicle. It may for example be a jeep-type vehicle, approximately 5 meters long and 1.8 meters wide. In this case, the antenna can be mounted on the roof of the vehicle, and the IDU is installed in the rear of the car. The antenna typically has a diameter of 1.2 meters. The HPC is typically a 100-watt TWT, and the LNC has a typical noise temperature of 80K. The IDU equipment is essentially identical to the IDU used in the portable earth station.

Table 2.4.5 – Transportable DSNG earth station main transmission parameters

	Transmission Mode			
	Normal	**Threshold Extension**	**Medium Quality**	**High Quality**
Video Coding Rate (Mbps)	7	5	10	13
Transmission Rate (Mbps)	11.3	13.3	13.3	17.7
Carrier Bandwidth (MHz)	8	7	8	10.6
Convolutional FEC Rate	3/4	1/2	7/8	7/8
No. of Carriers/36MHz Transponder	4	4	4	3
Threshold C/N at BER $1 \times 10\text{-}8$ (dB)	6.8	3.6	8.8	8.8

2.4.2 Technology for Mobile GSO Satellite Systems

From the beginning of commercial satellite communications (1965), the benefits of using satellites in geostationary orbits have outweighed the disadvantages. One of these disadvantages is the enormous free space propagation loss between the earth station and the satellite. Because of this disadvantage, earth stations have traditionally tended to be rather large installations, and only in recent years has the satellite communications technology matured to the point where it is now practical to use hand-carried portable terminals with geostationary satellites.

Inmarsat, the international organization with 79 member countries charged with providing maritime satellite communications, has led the way in providing land and air mobile communications. The equipment standards set up by the Inmarsat organization for accessing their satellites largely reflect the progression of technology and the physical size of the terminals. The ultimate goal, which will be realized with geostationary as well as non-geostationary satellites, is a handheld global communications terminal that fits into a pocket.

The Inmarsat-A terminal is still the most widespread communications terminal for mobile communications. It uses analogue technology and can support high-quality direct dial telephone, telex, facsimile and data services (9.6 and 64 kbit/s). Portable versions of this standard equipment fit into one or two suitcases and use folding antennas.

The Inmarsat-B terminal is the successor to the A-terminal and provides similar services. It uses digital technology and is therefore smaller, lighter and cheaper to buy, and has reduced user charges. Data services are provided up to 64 kbit/s.

The Inmarsat-C terminal is a small (a few kg) two-way data terminal and is the only terminal with an omni-directional antenna. It may be used in fixed, mobile, transportable, maritime and aeronautical versions. It supports store-and-forward message, text and data reporting communications at a rate of 600 bit/s.

The Inmarsat-M system also uses digital technology and provides direct-dial telephone, facsimile or 2.4 kbit/s data services. The physical size of Standard-M terminals has already progressed from the size of a small suitcase (11 kg) to the size of a notebook computer (about 2.6 kg). Further details on equipment used in the Inmarsat-M system are provided in Annex B.

2.4.3 Non-GSO Satellite Systems

Non-GSO systems require many satellites in order to provide uninterrupted service, with new satellites continually appearing over the horizon to take the place of those that are sliding out of sight on their way around the planet. The orbits of the various satellites in the constellation are designed to provide full global coverage. The non-GSO systems may occupy low earth orbits or medium to intermediate earth orbits.

Satellite constellations are tailored to provide specific regional or global coverage. Globalstar is an example of applying spacecraft constellation geometry and astro-dynamics to achieve a full-earth service area.

Since non-GSO satellites are closer to the earth than GSO satellites, they can achieve their function with smaller antennas and lower power per communications cell. The non-GSO satellites also enjoy less delay as signals from the earth travel and back to the ground have a shorter propagation delay.

A number of complex issues arise however e.g. the limited duration of visibility to a tracking, telemetry and command earth station. Ground resources, including spacecraft control and telemetry support, must be scheduled for contact times as these moving satellites come into view of an earth station antenna.

The advancement of technology in all fields used by satellite communications has resulted in increasingly more powerful and sensitive satellites, and shrinking physical size of earth stations. It was perhaps inevitable that the “earth station“ eventually would be reduced to the size of a hand-held telephone set. This is still not possible by using satellites in geostationary orbits due to the very large propagation path length resulting in signal attenuation loss and delay.

In order to overcome the disadvantages of the geostationary orbit satellites, communications systems may be implemented by using low earth orbit (LEO) or medium earth orbit (MEO) constellations. Technology at the end of the 20th century has matured to the point where such systems may be implemented with small hand-held earth stations. In fact, hand-sets will be designed for both terrestrial and satellite use with automatic switch-over between the two systems.

Orbit Constellations

If a LEO orbit constellation is used, the satellite altitude is selected much lower, typically between 700 and 2000 km. The orbit period in this case will be somewhere between 100 and 120 minutes.

Due to the lower altitude, the view of the world from the satellite is rather small and the number of planes to cover the entire globe will have to be increased to 6-8, and the number of satellites must be 6-12. The number of orbiting satellites, therefore becomes rather large.

Space Segments

The satellites may be implemented as simple repeater amplifiers with frequency conversion or they may have on-board signal processing capabilities.

The design life of the satellites obviously has an impact on the operating costs of any system implementation. The LEO orbiting satellites must be replaced more often than the MEO orbiting spacecraft, and the span of lifetime, therefore, may be in the order of 5-15 years.

In order to produce adequate power on the ground, satellites will be implemented with a large number of spot beams. This calls for rather complicated synchronization and “Handing-off” procedures by the controlling gateway stations, since a telephone subscriber while on conversation may find himself illuminated by 2 or more sweeping satellite beams.

Frequency Plan

Communications links between gateway stations and a satellite will use frequencies in the C or Ku bands. Furthermore, Ka bands are also used around 30 GHz for the up-links, and 20 GHz for the down-links.

As some of the frequency bands used are shared with GSO systems, there is a need for a balance between protecting the existing GSO services, and allowing new non-GSO systems to operate without undue constraint. Further agreements were reached at the World Radiocommunication Conference 2000 in Istanbul including some limits on earth stations of GSO networks, and power limits for non-GSO systems which define the rules of sharing the Ku band (10-18 GHz)

User Terminals

There will be a variety of user terminals employed in the LEO/MEO satellite systems. The bulk of user terminals will be small hand-held telephone sets that in many cases will be compatible with terrestrial cellular standards. The initial users will probably be international travellers who will be able to get a dial tone from anywhere in the world. Other subscribers such as users in remote areas where there is currently non-existent or very poor telecommunications infrastructure will follow.

Other terminals such as fixed terminals in village telephone booths, aircraft, ship or private residences will be used as well. Furthermore, multi-line units with several users sharing a single terminal will be developed. In most cases user terminals will be capable of interactive voice and voiceband data communications, such as facsimile, PC communications, and in some cases video conferencing.

Non-interactive communications in the form of world-covering one-way pagers that will be able to alert users and display alpha-numeric messages will also be available.

2.4.3.1 System Description

In the Annexes to Chapter 2.4 a number of LEO/MEO satellite communications systems are presented in some detail. Other systems are under preparation, and the inclusion of the above systems should not be seen as an endorsement, but rather as an attempt to illustrate representative system configurations. The description and details presented for each system is not exhaustive. Thus, the inclusion of certain equipment and functions for one system does not necessarily mean that the same equipment and functions are not included in the other systems, even if not described.

2.4.4 Global Mobile Personal Communications by Satellite (GMPCS)

The term Global Mobile Personal Communications by Satellite (GMPCS) refers to all communications systems that provide services directly to end users from a constellation of satellites. GMPCS represents the next stage in the development of wireless telephony, integrating the long-distance capabilities of satellites with the mobility of cellular phones. It accomplishes this by incorporating within portable user terminals all transmitting and receiving functions traditionally fulfilled by fixed earth station gateways and replacing terrestrial wireless networks with satellites. GMPCS uses geostationary and non-geostationary, regional and global, fixed and mobile satellite systems to interconnect individual user terminals. GMPCS systems are thus capable of transmitting any type of telephone transmission, i.e. voice, data, fax, or paging, to a destination anywhere in the world.

GMPCS thus embraces many of the systems and services addressed in this Chapter.

Further information is available in the ITU GMPCS Reference Book, December 1999.

2.4.5 Global Satellite Positioning Systems

Global Satellite Positioning Systems use satellites to provide highly accurate positioning data for a wide range of applications such as navigation on land, in the air, at sea and in space, national security, and new consumer oriented position determination applications.

The World Radiocommunications Conference 2000 provided additional allocations for radio-navigation satellite services. The additional spectrum makes it possible for the two current systems, Russia's Global Navigation Satellite System (GLONASS) and the USA Global Positioning System (GPS) to develop into more accurate second generation systems, while providing room for Europe's new system, Galileo.

2.4.6 ITU-R Publications

The ITU-R prepared following Handbook and Supplements dealing with Satellite communication:

- Satellite Communications (Fixed-Satellite Services), 2nd edition 1988
- Supplement No. 1 to Handbook on Satellite Communications: Fixed-Satellite Service, 1995
- Supplement No. 2 to Handbook on Satellite Communications: Computer Programs for Satellite Communications
- Supplement No. 3 to Handbook on Satellite Communications, VSAT Systems and Earth Stations, 1995
- Radiowave Propagation Information for Predictions for Earth-to-Space Path Communications. 1996.

2.4.7 Abbreviations

AC	Assignment Channel (or) Alternating Current
ACU	Antenna Control Unit
AFC	Automatic Frequency Control
AGC	Automatic Gain Control
AM	Amplitude Modulation
AOR	Atlantic Ocean Region
BER	Bit Error Rate
C/N	Carrier-to-Noise
CSMS	Computerised Station Management System
DAMA	Demand Assigned Multiple Access
DC	Direct Current
DCME	Digital Circuit Multiplication Equipment
DOMSAT	Domestic Satellite
DSI	Digital Speech Interpolation
DTMF	Dual Tone Multi-Frequency
ESC	Engineering Service Channel
FEC	Forward Error Correction
FET	Field Effect Transistor
FM	Frequency Modulation
GaAs	Gallium Arsenide
GSO	Geostationary Orbit
HDTV	High Definition Television
HIC	Hybrid Integrated Circuits
HPA	High Power Amplifier
IDR	Intermediate Data Rate
IDU	Indoor Unit
IF	Intermediate Frequency
IOR	Indian Ocean Region
ISC	International Switching Centre
ISDN	Integrated Services Digital Network
LCD	Liquid Crystal Display

LEO	Low Earth Orbit
LES	Land Earth Station
LNA	Low Noise Amplifier
LNC	Low Noise Converter
LRE	Low Rate Encoding
LSI	Large Scale Integration
MC	Multi-Clique
M&C	Monitoring and Control
MD	Multi-Destination
MEO	Medium Earth Orbit
MHIC	Microwave Hybrid Integrated Circuit
MIC	Microwave Integrated Circuit
MPEG	Moving Picture Expert Group (ISO)
MTBF	Mean Time Between Failure
NMS	Network Management System
ODU	Outdoor Unit
PC	Personal Computer
PCM	Pulse Code Modulation
PDH	Plesiochronous Data Hierarchy
PIN	Personal Identification Number
POR	Pacific Ocean Region
PM	Phase Modulation
PSK	Phase Shift Keying
QPSK	Quadrature Phase Shift Keying
SD	Single Diversity (or) Space Diversity
SDH	Synchronous Data Hierarchy
SIM	Subscriber Identity Module
SSPA	Solid State Power Amplifier
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TSI	Time Slot Interchange
TV	Television
TWT	Travelling Wave Tube
VSAT	Very Small Aperture Terminal

ANNEX 2A

VSAT System

The following is a detailed description of one particular implementation of a VSAT System.

The Hub or Control Station

The Hub VSAT equipment may be constructed as a new earth station or it may be integrated into existing earth station equipment at the GCE level. For example, the existing equipment at the earth station may include a 13-meter antenna, Low Noise Amplifiers (LNA), High Power Amplifiers (HPA), Up-converters and Down-converters, Multiplexers (MUX), RF combiners, RF dividers, and 140 Mbit/s digital radio link to the International Transmission Maintenance Centre (ITMC).

The equipment to be added to the Hub station for VSAT operations is the following subsystems:

- Up-converters and down-converters for the applicable C- or Ku frequency bands.
- Voice and data channel modems. These may be plug-in cards for continuous voice and data communications with various data transmission rates. For example, the voice modems are typically able to operate at 16 and 32 kbit/s information coding rates, and the data modems at various rates between 9.6-2048 kbit/s. Each individual modem can be set to the desired transmission rate (Bandwidth-on-Demand function). It should be noted that the duration of a continuous type of communications channel in contrast to an interactive type of communications does not necessarily mean a permanently assigned communications channel. It may be assigned for the duration of a telephone call, forever, or for any length of predetermined time desired.
- The DAMA subsystem, in which the satellite transponder circuits are assigned on demand for voice or data communications either between the Hub and VSAT stations (star configuration), or between individual VSAT terminals (mesh configuration). The DAMA subsystem is communicating with remote VSAT terminals on a continuous TDM (Time Division Multiplex) Common Signalling Channel (CSC) (outbound or transmit communications), and by Time Division Multiple Access (TDMA) bursts from the VSAT terminals (inbound or receive communications). Modern DAMA equipment can control thousands of VSAT terminals.
- The AA/TDMA (Adaptive Assignment/Time Division Multiple Access) subsystem in which rapid and frequent interactive communications is implemented with the remote VSAT terminals. The modes of communications with the remote VSAT terminals are the same as for DAMA operations, i.e. TDM outbound and TDMA inbound.
- Data Voice Multiplex (DVM) equipment, in which the VSAT communications channels are interfaced to the existing digital MUX equipment. This interface is done at 2048 kbit/s to the existing MUX, but at various data rates to the VSAT Hub equipment.
- The Network Management System (NMS), from where the entire VSAT network can be managed, monitored and controlled. The NMS consists of a powerful workstation computer with applicable software for VSAT System Monitoring and Control, VSAT User Monitoring and Control, and for Billing management.

Voice Channels at the Hub Station

The voice channels may be configured from the NMS for either 16 or 32 kbit/s information rate. Furthermore, this information rate may be reduced to 8 kbit/s in the future without any hardware change. Implementation to 8 kbit/s is done simply by downloading information from the NMS to the hub station equipment and individual VSAT terminals. The 16 or 32 kbit/s voice channels at the hub station are input and output to/from the BOD voice channel modems as analogue channels.

At the ITMC, similar DVM equipment as at the Hub station is implemented for the analogue/digital conversion between the analogue telephone channels from the telephone exchange and the existing digital radio MUX equipment. The interface here is also at 2 Mbit/s. In the voice operations mode, one unit of DVM equipment is typically capable of handling a maximum of 30 voice channels.

Data Channels at Hub Station

The BOD data channels may also be configured from the NMS for data channel rates between 9.6 and 2048 kbit/s. Since the interface at the hub station to the existing digital radio link MUX equipment is done at 2 Mbit/s, rate conversion is required. This rate conversion is done in the Data Voice MUX equipment mentioned in the previous paragraphs.

The DVM equipment has 5 plug-in cards. For voice operation, each card can handle 6 voice channels for a total of 30 voice channels. For data channel operations, depending on the data rate, each card can handle 2-4 channels per card. The DVM can operate with a mixture of voice and data channels, but in no case can the aggregate bit rate from all input channels exceed 2048 kbit/s, which is the output data rate.

Interactive AA/TDMA Data Channels at the Hub Station

The interactive, frequent bursts of information to be transmitted to remote VSAT terminals are coming from a host front-end computer which is located at the premises of the user or owner of the VSAT system. The TDM data channel used for the outbound communications is typically operating at 64 or 128 kbit/s, and the return burst of information from the VSAT terminals are typically coming at 64 kbit/s. Since the host computer is not at the earth station (or even at the ITMC), multiplex equipment must also be included at the earth station and the ITMC to handle the interactive AA/TDMA channel.

VSAT Terminals

Depending on the required frequency bands and transmission capacity of the remote VSAT stations, antenna sizes of 1.2, 1.8, 2.4 and 3.7 meters are typically used. The Low Noise Converters (LNC) at C-band may have a 60 K noise temperature, and 140 K at Ku-band. High Power Converters (HPC) at C-band may use Solid State Power Amplifiers (SSPA) of 3, 10 or 20 watts rated output power, and similar devices for the Ku band may be rated at 1, 2 or 6.6 watts. The size of the antenna, and the LNC and SSPA ratings for a certain terminal, are not only dependent on the number of transmitted carriers, but also on whether the carriers are directed to the Hub station only (star configuration) or to other VSAT terminals (mesh configuration). If most of the carriers are transmitted to the Hub station, the size of the VSAT station can be minimized, or alternately, the number of carriers can be increased. This will also directly influence the power consumption and the cost of the VSAT terminal.

VSAT terminals, particularly if they are used for rural communications, are often implemented with solar power plants. Table 1 shows the C-band transmit capability and power consumption of various terminals operating with an Intelsat-7 type satellite.

Table 1 – Terminal C-Band transmit capabilities

ANT	ODE	No. of channels	Prime power consumption
1.8 M	5 W	5	337 W
1.8 M	20 W	5	500 W
2.4 M	5 W	5	337 W
2.4 M	20 W	10	700 W
3.7 M	5 W	5	337 W
3.7 M	20 W	10	700 W

The VSAT terminals may be implemented initially with only the antenna, the ODE and the Main Indoor Unit. The IDU is a container that may be placed on a tabletop and has no particular installation requirements. It contains up to 5 plug-in voice/data modems, of which one is used as the Common Signalling Channel (CSC) for communications with the Hub station. The Main IDU may be expanded with a total of 2 sub-IDUs for a total of 15 plug-in modems (including the CSC modem).

The Main IDU also may be expanded with the AA/TDMA Indoor Unit for interactive communications with the Hub station. The major components in this IDU are the Satellite Channel Interface (SCI) unit (modem, frequency synthesizer), a Baseband Processor (BBP) interface unit, an Operations Panel (OPP), and a power module.

Although the VSAT terminals are generally considered to be low-capacity earth stations, the capacity for each terminal is limited not so much by the equipment, as by the transmitted power, or alternately by the receive sensitivity of the satellite transponder. If the 15 channels in one VSAT terminal are implemented with data modems and each set to 2048 kbit/s, the number of 64 kbit/s channels that theoretically can be transmitted is 450 channels. If these 64 kbit/s channels were to be used with Low Rate Encoding (LRE) equipment, which is currently available with acceptable quality at 16 kbit/s, one VSAT terminal would theoretically be able to transmit 1800 channels.

Analogue or Digital TV Distribution from the Hub Station

Analogue or digital television may be transmitted from the Hub station by using the same RF equipment as the VSAT channels at the Hub station and at the remote VSAT terminals. However, if analogue TV is used, the VSAT antenna size may have to be increased. As an attractive alternative, digital TV services can be included economically because the antenna sizes as outlined in this example may be used for the VSAT terminals with no modifications being required to the terminals. A TV receiver is connected directly to the IDU at 1 GHz IF level.

Network Management System (NMS)

The VSAT network is controlled technically as well as managerially from the Hub earth station, although the NMS functions can be performed remotely from some other location (for example the ITMC).

The main functions performed by the NMS can be broadly categorized as follows:

1) Network Status Monitoring.
2) Network Configuration Management.
3) Network Control.

4) Network Reliability (switchover functions).

5) Alarm Management.

6) Statistics Management.

7) Logging Management.

Power Generating Equipment

The VSAT terminal equipment is very compact and requires no special shelter. The ODE unit is installed directly on the antenna structure, and the IDU cases may be located at any convenient place up to 80 meters from the antenna. This distance may be extended if low resistance cables are used.

The IDU will operate normally up to a temperature of 40 °C, and it is therefore not necessary to provide air conditioning except in extreme cases. If the prime power source is unreliable or not available, the VSAT terminals can be operated by solar generating power plants. Typical power consumption of the VSAT equipment was shown in a previous Table. Alternately, a small mobile diesel generator may be used.

ANNEX 2B

The Inmarsat system

The Inmarsat A, B, C and M standards are used for equipment in maritime or land-based services. Additional equipment standards for aeronautical services are the following:

- Aero-C for store-and-forward text and data messages to be sent and received by aircraft anywhere in the world.
- Aero-L for low-gain real-time data-only communications service for flight-deck and airline operations purposes.
- Aero-H for high-gain service providing multiple-channel flight-deck voice and passenger telephony.

Additional Inmarsat standards in preparation are Inmarsat-D, which is a pager system, and Aero-I which is a mobile voice and data service using smaller, lighter and less expensive avionics equipment and antennas designed for short- and medium-haul aircraft.

Calls originated from any of the standard equipment in the Inmarsat system must be routed through a Land Earth Station (LES), sometimes called coast earth stations or aeronautical ground earth stations. Numerous LESs are located worldwide in the Atlantic Ocean Region (AOR), Indian Ocean Region (IOR) and Pacific Ocean Region (POR).

It should be noted that the Inmarsat system is not designed to provide direct service from one mobile terminal to another. Rather, it is intended for a mobile terminal to communicate via a LES with subscribers connected to a public switched telephone or data network. Hence, if a communications link is set up between two mobile terminals, it will suffer a satellite double hop propagation delay which is unsatisfactory for telephone connections, but may be of little significance for data communications links.

If the purpose of rural communications is to establish voice and/or data links with urban centres that are already well connected to an LES, the Inmarsat-M system may currently be the fastest and least expensive way to integrate rural communities into an existing public communications network for purposes of trade, telemedicine, personal relationships and a host of other communications needs. If, however, the purpose is primarily for intercommunications between locations within or between rural communities, other communications architectures may be more appropriate.

A Portable Terminal for the Inmarsat-M System

The Inmarsat-3 series of satellites will eventually comprise 5 spacecraft covering all ocean areas. The transponders on the Inmarsat-3 satellites when used with spot beams are powerful enough to allow small notebook-size terminals to operate with an LES station and thus be connected to the international public telecommunications network.

The terminal uses the standard Inmarsat transmit frequency band of 1626.5 to 1660.5 and receive band of 1525.0 to 1559.0 MHz. Voice channels are available and encoded at 4.8 kbit/s by using advanced multiband excitation techniques. Facsimile may be transmitted and received using the G3 standard at 2.4 kbit/s, and asynchronous data transmission is available up to 2.4 kbit/s.

A built-in rechargeable lithium ion battery is used to power the equipment. When fully charged it will power the terminal for 4.5 hours in receive mode and for about 1.2 hours in transmit mode. It may be recharged from a 90-260 VAC or 12 VDC source.

Operating the terminal is fairly simple. The planar antenna, is pointed towards the satellite location for the ocean area where the terminal is located. By depressing the Menu button on the handset, the built-in LCD display on the handset will display the signal strength of signals coming from the satellite. By reorienting the antenna, the terminal may be set to receive maximum signal strength. If the direction to the satellite is unknown, a compass will aid the user in the initial orientation. An icon display will indicate that the terminal is locked on to a satellite.

The Base Unit of the Portable Terminal

The base unit of the portable terminal consists of the transmit and receive electronics, the antenna, the rechargeable battery and the cradle for the handset. The handset cradle may be detached and connected to the base unit by a cable up to 4.5 meters long so that the user of the terminal may be at his workplace while the base unit and antenna may be placed near a window. Furthermore, the antenna may also be detached and placed up to 10 meters from the base unit.

The handset cradle also serves as a connection point for a fax machine, an external telephone or a personal computer.

The base unit also contains a so-called SIM (Subscriber Identity Module) card reader. The SIM card is a “smart” card (about the size of a credit card) with an embedded electronic chip which provides enhanced security and many value added features such as:

1) personalized electronic phone book which can be easily edited and used for speed dialling and other services,
2) PIN (Personal Identification Number) code for restricting the use of the terminal,
3) personal ID providing a single number regardless of the location of the terminal,
4) security key and secret authentication algorithm for fraud protection,
5) language preference for LCD screen messages,
6) call barring capabilities for restricted dialling, and
7) short text messages for two-way communications.

SIM cards gives the user convenience and mobility. When a SIM card is used, the terminal senses the number and automatically bills call charges to a predetermined account.

The Handset of the Portable Terminal

Except for the insertion of the SIM card, the operation of the Inmarsat-M terminal is performed from the handset. Despite its small size, numerous functions are built in to this intelligent handset. The main functions are listed next:

- *Storing Telephone Numbers*

  Users can store names and numbers in the handset and access them as an electronic directory. Numbers stored on the SIM card can also be accessed.

- *Battery Status Indication*

  A four-stage icon on the LCD indicates the battery status. An alarm and LCD messages alert the user to low battery condition before the end of talk time.

- *Call-in-Absence Indication*

  Un-answered calls are indicated by an LCD icon. The last 10 calls are displayed with caller ID and times and dates if the calls were made via ISDN.

- *Credit Card Calling*

  Users can make credit card calls by keying in a two-digit prefix and the credit card number.

- *Call Duration Display*

  Call duration is displayed at the end of every call. The handset also displays the duration of the previous call.

- *Signal Strength Indicator (visual and aural)*

  Signal strength levels are displayed on the LCD. If the Menu button is pressed, the current signal strength is shown. Users can also set the terminal to emit beeping tones to indicate the signal strength.

- *Barred Calls*

  The user can set the terminal to prevent calls to specified numbers or countries.

- *Other Functions*

  Power save mode, backlight, redialling, user's numbers/ocean region display, phone lock, speed dialling, mute.

ANNEX 2C

The globalstar system

General

The following information is extracted from material provided by courtesy of Globalstar L.P.:

The Globalstar system consists of a Space Segment, a User Segment, a Ground segment, and a Terrestrial Network as shown in Figure 1. The Globalstar system provides communications from any point on the earth surface to any other point on the earth surface, exclusive of the polar regions. The satellite orbits are optimized to provide highest link availability in the area between 70 degrees south latitude and 70 degrees north latitude. Service is feasible in higher latitudes with decreased link availability.

User Terminals can be served by a satellite 10 to 15 minutes out of each orbit. A smooth transfer process between beams within a satellite and between satellites provides unbroken communications for the users. The orbital planes are inclined at 52 degrees. Coverage is maximized in the temperate areas with at least two satellites in view, providing path diversity over most of the area. There is some small sacrifice in multiple satellite coverage at the equator and at latitudes above 60 degrees.

The Gateways to the terrestrial network are illuminated by an earth coverage beam. The Gateway connects the User Terminal to the terrestrial network via an earth terminal. The terrestrial network is not a part of Globalstar.

Globalstar can locate the position of a User Terminal and provide the location as a service. Accuracy of the position location service is a function of several variables including (a) Number of Satellites in View, (b) Position Accuracy of the Satellites, (c) Geometry of the User Terminals, Satellites and Gateways, (d) The length of time that the User Terminal is connected to the Gateway.

The Gateway will also support more precise User Terminal position location as a service. With clear line of sight to at least two satellites, separated by at least 22 degrees as seen by the User Terminal, the Gateway will be able to compute the User Terminal’s position to within 300 meters with 95% probability in less than 10 seconds.

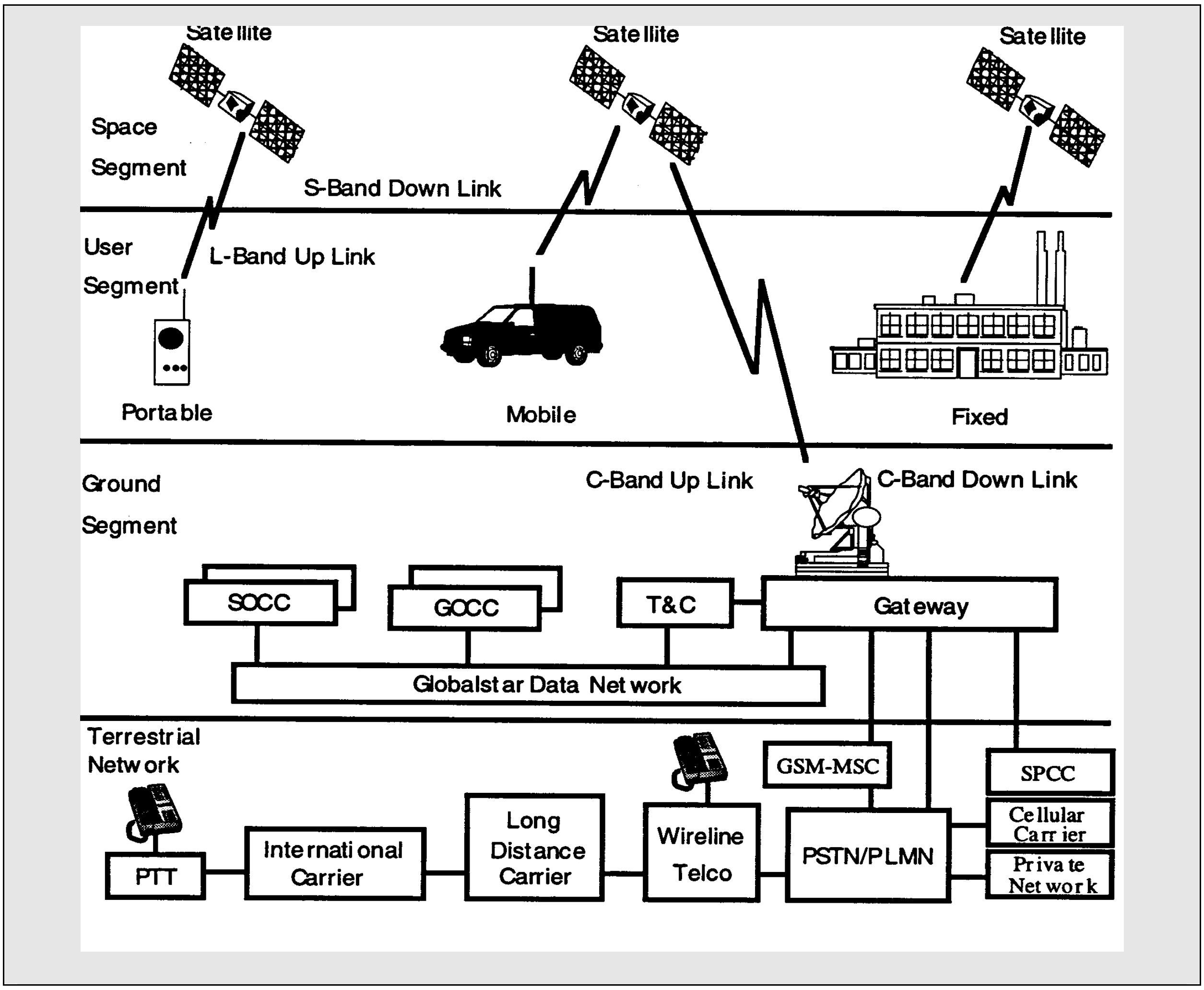
Space segment and orbit constellation

The Globalstar space segment consists of 48 satellites in 1410 km Low Earth Orbits. The low orbits permit low power handsets similar to cellular phones. These satellites are distributed in 8 orbital planes with 6 equally spaced satellites per orbital plane. Satellites complete an orbit every 114 minutes. User Terminals in a particular location on the surface of the earth are illuminated by a 16 beam satellite antenna as it passes over the earth.

The Globalstar satellite is a simple low cost satellite designed to minimize both satellite costs and launch costs. A User Terminal transmits to the satellite by L-Band. The signal enters the satellite through the L-Band low noise amplifier. It is amplified and then converted into a C-Band signal after which it is further amplified. This is radiated to the Gateway. The Gateway receives the signal and block downconverts to an intermediate frequency. A sample of the intermediate frequency is provided to the TCU for processing. The communications traffic is presented to the CDMA equipment for demodulation.

In the transmit direction, the Gateway combines the uplink CDMA signals with the signal from the command transmitter and radiates it at C-Band up to the satellite. The satellite then down converts the signal and radiates an S-Band downlink signal to the User Terminals. Table 1 shows typical link budgets for all frequencies used by the Globalstar system.

Figure 1 – Overview of the Globalstar Leo System



Telemetry and commanding shares the C-Band with the communications feeder links. The T&C connects to the normal C-Band communications antenna for on-orbit operations. There is also a T&C antenna on the anti-earth face of the satellite. This antenna functions when the satellite is not oriented correctly or when there are problems. The anti-earth antenna is to insure that telemetry can be obtained and commands entered under all recoverable contingencies. Note that the anti-earth antenna bypasses the Low Noise Amplifier. This means that the transmitted power from the commanding earth terminal will have to be higher than the normal power required for commanding. This can be accommodated because there is no communications traffic when the satellite is not oriented correctly.

The satellite is steered in yaw to keep the solar panels oriented toward the sun to extract the maximum energy. This increases the communications capacity of the Globalstar System. There are some minor penalties that cause a slightly slower acquisition time and may cause more hand offs than would be otherwise required.

Table 1 – Typical link budgets for all frequency bands

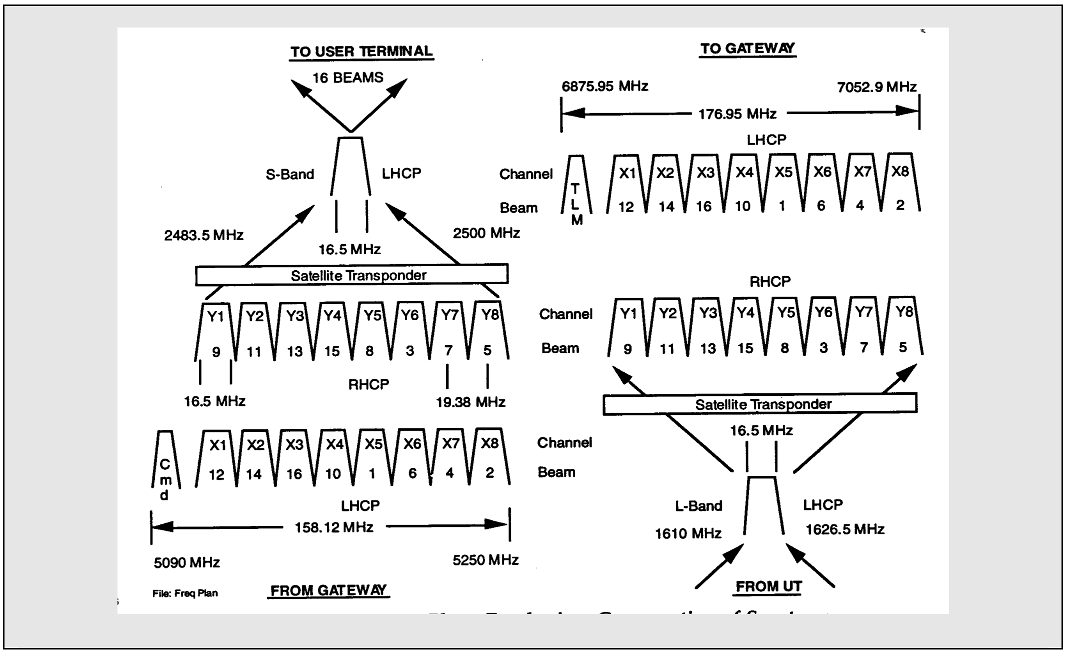
Parameter	Forward link		Return link		Units
	Satellite to user terminal	Gateway to satellite	User terminal to satellite	Satellite to Gateway	
Nominal frequency	2.5	5	1.6	7	GHz
EIRP/User (nadir, clear)	1.1	36.4	–14.3	–33.3	dBW
Space loss (nadir, clear)	163.4	169.7	159.6	172.3	dB
Other factors	1.4	7.90	2.10	1.60	dB
G/T (hand held)	–26	–29.6	–14.25	27.5	dB/K
Eb/No.*	5.1	24	4.5	15.1	dB
Average data rate	2400	2400	2400	2400	bps
Diversity benefit			2.2		dB
Composite Eb/No.	5		6.3		dB

* Satellite to U.T. & Satellite to gateway link Eb/No are combined results of both up and down paths.

Frequency plan

The frequency plan adopted by Globalstar is illustrated in Figure 2.

Figure 2 – Globalstar frequency plan



LHCP = Left-hand Circular Polarization,
RHCP = Right-hand Circular Polarization,
UT = User Terminal

Ground segment

Gateways

The Gateway interconnects the Globalstar satellite based wireless network and the PLMN, such as AMPS and GSM, or directly into the local telephone office (PSTN). As such, it is a termination point for network transmission and network signaling. The Gateway can be connected to existing PSTN using standard El/T1 trunk supporting a variety of signaling protocols. To GSM networks, the Gateway appears as a GSM Base Station Subsystem. To those mobile switches in the EIA/TIA environment, it appears as another mobile switch supporting the IS-41 Intersystem Operation Standard. In all cases, inter-operability between Globalstar and telephone/cellular companies is assured and the subscriber maintains a convenient single point for billing.

The Gateway is designed in a modular manner to provide flexibility in order to grow in step with market demand. The Globalstar Gateway can be shared by multiple service providers who can share investment in start-up common equipment. This design provides security from other service providers sharing the gateway and expansion is planned as revenues rise.

Each gateway will contain up to four tracking antennas and radio frequency front ends that track the satellites orbiting in their view. Gateways are expected to cost from $2 million to $5 million, depending upon the number of subscribers being serviced by the gateway and assuming that the gateway will be located at the site of an existing cellular or other appropriate telecommunications switch. Each nation with at least one gateway within it borders will have complete control over system access by users within its territory. A single gateway is expected to be able to provide fixed coverage over an area larger than Saudi Arabia and mobile coverage over an area almost as large as Western Europe. Thus, full global land-based coverage of virtually all inhabited areas of the globe could theoretically be achieved with as few as 100 gateways. Globalstar believes, however, that as many as 210 gateways may be required to minimize land-based long-distance charges and to respect national boundaries.

Although Globalstar has commissioned the design of the gateways to be used with the Globalstar system, ownership and operation of the gateways will be the responsibility of service providers in each country or region in which Globalstar operations are authorized.

Satellite Control

Globalstar's SOCCs will track and control the satellite constellation using command and telemetry units located in various gateways around the world and telemetry stations in two locations in the United States. The SOCCs will control satellite orbit positioning, maneuvers and station keeping, and will monitor satellite health and status in all subsystems.

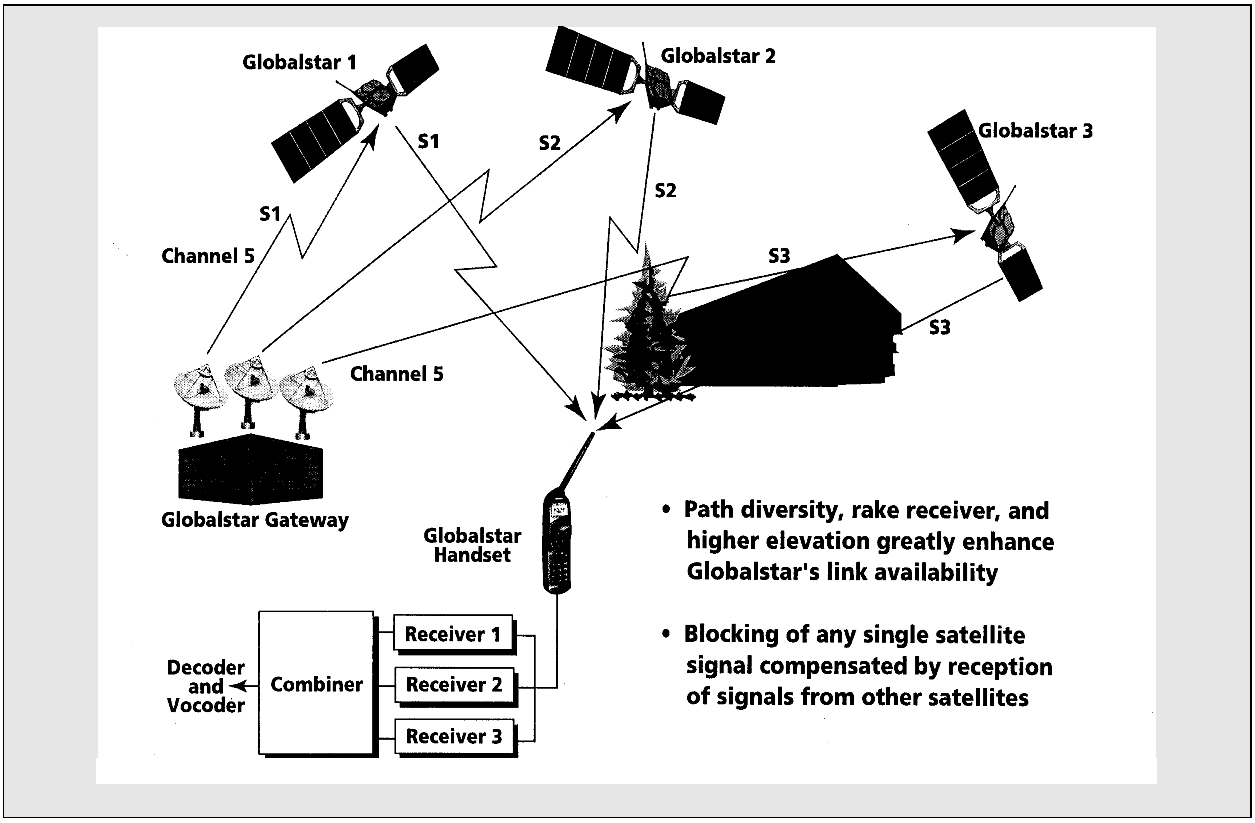
Furthermore, Globalstar has entered into a contract with QUALCOMM Inc., providing for the design, development, manufacture, installation, testing and maintenance of two ground operations control centres (GOCCs). The GOCCs, which will be in continuous operation 24 hours a day, will be responsible for planning and controlling satellite utilization by gateway terminals and coordinating information received from the SOCCs. In addition to these planning functions, the GOCCs will be responsible for monitoring system performance, collecting information for billings to service providers and ensuring that the gateways do not exceed allocated system capacity. The GOCCs will be responsible for dynamically allocating system capacity among nearby regions to optimally service changing patterns of demand.

User Terminal

- *Dual-mode User Terminal*

The Globalstar/GSM Dual-mode User Terminal offers a global roaming solution for GSM cellular phone users. Globalstar enhances wireless service options by providing worldwide GSM cellular-like services in areas outside traditional cellular coverage. Inside the GSM service area, a Globalstar subscriber can continue to use the GSM services. Outside the GSM service area subscriber may roam to other GSM cellular networks or switch to the Globalstar network. Even in areas not served by cellular companies, subscribers are ensured of reliable, cost effective, world class mobile communication by Globalstar. The Globalstar constellation of Low Earth Orbit satellites will provide digitally crisp voice, data, and facsimile services. In all cases, inter-operability between Globalstar and GSM is assured and subscribers maintain a convenient single point for billing. For the mobile subscribers, a vehicle mount car kit is available. The Globalstar handheld unit can be mounted in a cradle that provides power to extend battery life and allows for handsfree operation for convenience and driving safety.

Figure 3 – Path diversity of the Globalstar system



Tri-mode User Terminal

The Globalstar/IS-95/AMPS Tri-Mode User Terminal offers a global roaming solution for AMPS/IS-95 cellular phone users. Globalstar enhances wireless service options by providing worldwide digital cellular-like services in areas outside traditional cellular coverage. Inside the cellular AMPS or IS-95 Code Division Multiple Access (CDMA) digital coverage area, the Globalstar subscriber could continue to use the land based services. Even in the areas not served by cellular services, the subscriber is insured access to reliable, cost effective, world class mobile communication available from Globalstar. The Globalstar system of low Earth orbit satellites will continue to provide digitally crisp voice, data, and facsimile services using hand-held, vehicle mounted or fixed products. In all cases, inter-operability between Globalstar and cellular companies is assured and the subscriber maintains a convenient single point for billing.

For mobile subscribers, a vehicle mounted car kit is available. The Globalstar handheld unit can be mounted in a cradle that provides power to extend battery life and a handsfree operation for convenience and driving safety.

Fixed Single Line

Globalstar's family of fixed products meet the immediate communication demands for voice, facsimile and computer transmissions, and provides the flexibility, capacity and quality needed for future growth in demand. Wireless access offers an innovative solution for quick and efficient installation of a communications link for "hard to reach" subscribers without the cost and time necessary to interconnect using traditional wire. Globalstar delivers a worldwide contiguous wireless solution for delivering the "last mile" of connectivity with the telephone network.

Globalstar has designed their fixed station products to allow local content and subscriber flexibility. A Globalstar antenna, Radio Unit and optional digital telephone is provided by Globalstar. This digital phone includes a display for call progress indicators, icons for voice mail and memory for those frequently call numbers. The subscriber equipment can be locally provided by the in-country service provider. The antenna is mounted at a convenient outdoor location with a clear view of the sky and connected to the subscriber equipment. Globalstar provides cost effective interconnection to the telephone office for the unserved "hard to reach" population in developed and under developed urban and rural areas.

Multiline Unit

Globalstar delivers a worldwide contiguous wireless solution for delivering the "last mile" of connectivity with the telephone network. Our family of fixed products give local operators the flexibility, capacity and quality needed for future growth. Wireless access offers an innovative solution for quick and efficient installation of a communications link for "hard to reach" subscribers without the cost and time necessary to interconnect using traditional wire.

The Globalstar Multiline unit emulates a standard trunk circuit for interconnection with switching concentrators such as a PABX. Subscribers can be connected to the PABX using traditional wire, wireless or fiber and communicate with each other within the PABX. Globalstar is accessed when communication is required outside the switching system. The PABX subscriber will dial the outside line access number and Globalstar will connect the calling party to the public telephone office or other remote switching centre.

ANNEX 2D

The ICO system

The following information is based on information provided by ICO:

General

ICO is a satellite-based mobile communications system designed primarily to provide services to pocket phones. The system will offer digital voice, data, facsimile, and a suite of messaging services anywhere in the intended environments world-wide. Figure 1 is a diagram of the overall ICO system configuration.

The system design integrates mobile satellite communications capability with the public land mobile networks (PLMNs) and employs, among others, handheld mobile telephones ('handhelds') which offer services similar to normal cellular phones, in outdoor environments. It will route calls from PLMNs and public switched telephone networks (PSTNs) through ground stations (called Satellite Access Nodes or "ANs" which will select a satellite through which the call will be connected. Calls from a mobile terminal will be routed via the satellite constellation to the appropriate fixed or mobile networks or to another mobile satellite terminal. Handsets will be produced by major manufacturers of telecommunications equipment, benefiting from terrestrial cellular/PCS technology. Single-mode satellite-only versions will be available, but most are expected to be capable of dual-mode operation with both satellite and terrestrial cellular/PCS systems. Dual-mode handsets will be able to select either satellite or terrestrial modes of operation automatically or under user control, subject to the availability of the satellite and terrestrial systems and the user's preferred service arrangements. A feature of the ICO system will be a suite of higher-penetration messaging functions capable of delivering an alert or short message.

Extensive measurements of the radio wave propagation typical of that expected to be experienced by handheld users operating to satellites, including the effects of shadowing and unwanted reflections, were made in a range of representative rural, suburban, and urban environments. The results were combined with the moving geometry of the satellite constellation to assess the service availability, expressed as the probability of a handheld user being able to initiate or receive a call at any given time.

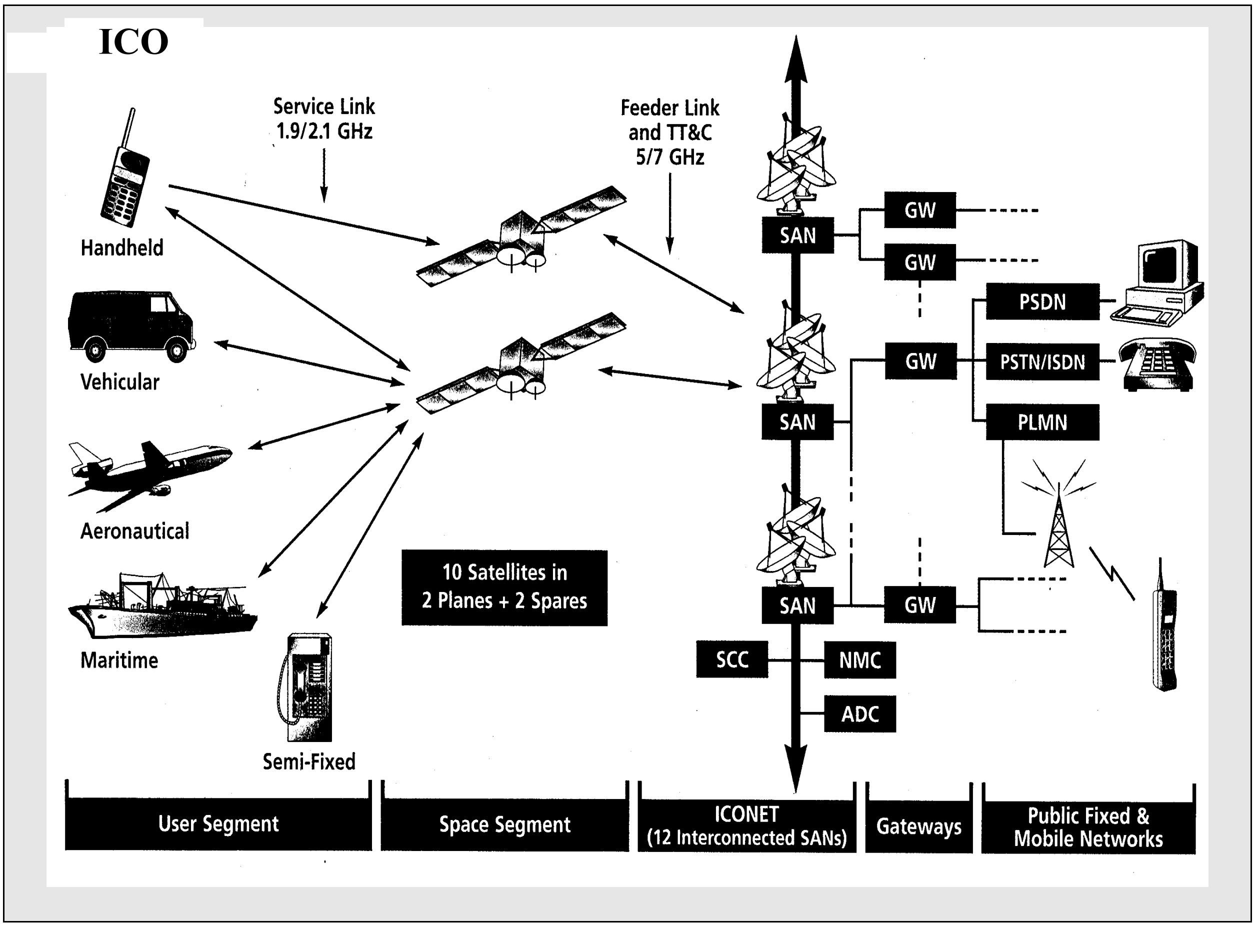
An average service availability of more than 90 per cent can be provided in rural and suburban outdoor environents by providing a link power margin of about 7dB and using path diversity. In these same environents, without path diversity, a link margin of about 16 dB would be needed to achieve comparable availability. These results confirm the benefits of path diversity as a system design philosophy in terms of limiting the transmitted power and receiver sensitivity needed by individual satellites in the constellation, and thus reducing their complexity.

Space Segment and Orbit Constellation

A constellation of 10 satellites in medium earth orbit (MEO), 10,355 km above the earth's surface will be arranged in two planes of five satellites each, with one spare satellite in each plane (i.e. 12 in orbit). Each orbital plane is inclined 45 degrees to the equator. 12 Satellite Access Nodes are located globally. Hughes Space & Communications International, Inc., will build the satellites under a contract signed in July 1995.

The configuration has been designed to provide coverage of the entire surface of the earth at all times and to maximise the path diversity of the system. Path diversity is the availability to a user of more than one satellite at the same time, and provides an alternative path for transmission in case one satellite is obstructed, increasing the likelihood of uninterrupted calls.

Figure 1 – Overview of ICO system configuration



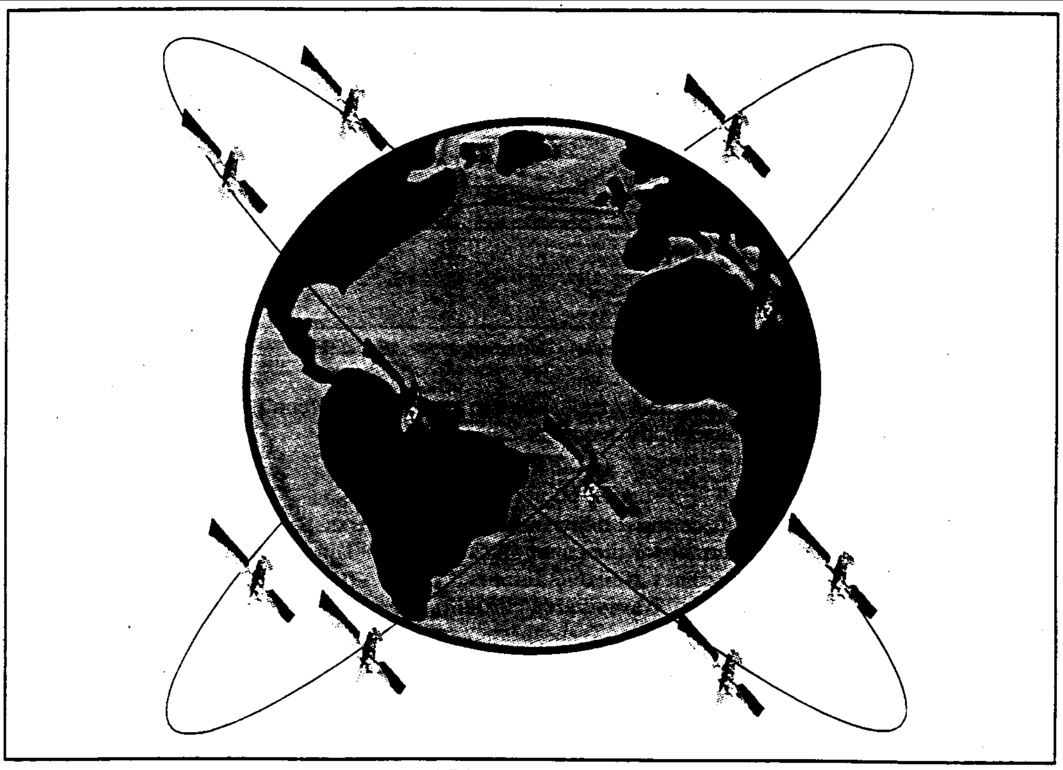
Satellite Constellation

The orbital pattern is also designed for significant coverage overlap, ensuring that usually two but sometimes three and up to four satellites will be in view of a user and a SAN at any time. Each satellite will cover approximately 30 per cent of the earth's surface at a given time. The satellite orbits have been selected to provide coverage of the entire globe on a continuous basis, while allowing high elevation angles to users, averaging 40-50 degrees. Figure 2 shows the ICO orbit constellation.

Satellite Design

The satellites use the proven and popular HS601 geostationary satellite design, with attitude and control systems modified to suit the special requirements of the MEO orbit. The communications payload is of transparent design, allowing flexibility to transmission format, using a high degree of digital technology for functions such as channelisation and beam generation that have traditionally been performed by analogue technology. The digital technology provides a very flexible satellite configuration (for example, flexible allocation of capacity across the full 30 MHz available in the 2 GHz service link bands), while having significant advantages over analogue technology in terms of production and manufacture for the comparatively large production run as compared with more conventional geostationary satellite orders.

Figure 2 – ICO orbit constellation



Inmarsat-P ICO-10 Satellite Constellation

Links between individual users and satellites will be established via service antennas mounted on each of the satellites. To provide robust radio links with handheld units, the satellites use antennas with an aperture in excess of two metres. The use of multiple service link beams on each satellite also allows frequency re-use and increases the efficient use of spectrum allocation.

The SANs are the primary interface with the satellite for coordinating and routing traffic. The SANs implement call routing to ensure the best quality of service to system users. In this regard, each SAN tracks the satellites within its line of sight and routes communications traffic to the optimal satellite.

Each satellite is designed to support at least 4,500 telephone channels distributed among approximately 750 carrier waves by using time division multiple access (TDMA) technology to integrate the multiple channels into each carrier wave. TDMA technology was selected after careful consideration of other technologies, including CDMA. Internal and industry studies for the ICO system have judged TDMA to have larger traffic-handling capacity and less demanding technical requirements than CDMA, and to be less subject to call failure as a result of interference. The design capacity of the individual satellites is such that this constellation will support at least 2,400 million voice minutes per year, assuming typical traffic distribution patterns with traffic "hot-spots". In addition, excess capacity will be available outside traffic "hot-spots" for further traffic opportunities. The life span of ICO satellites is expected to be approximately twelve years.

Satellite Technology

Service link and number of beams: The system performance will be well in excess of that required to provide the desired level of service. The 163 transmit and receive service link beams will provide links with a power margin in excess of 8 dB; the average link margin will approach 10 dB rising to 10-11 dB for a user at sub-satellite point.

Feeder link antennas: Feeder link antennas support the link between the satellites and the SANs. At any time, each satellite will usually be in direct contact with between two and four SANs. Before a satellite falls outside the line of sight of one SAN, it will establish contact with another SAN. This SAN will then track the satellite whilst it is in its line of sight and it will direct calls to this satellite or any other satellite within its sight depending on which one offers the best signal quality.

Satellite mass and power estimates: The total satellite launch mass, for a 'direct injection' into the circular orbit, is about 2600 kg allowing multiple launch vehicle capability. Direct injection allows some simplification to the HS601 as no apogee motor is to achieve final orbit. The solar arrays will use the latest Gallium Arsenide cells to provide end of life powers in excess of 8,700 W.

Selection of Orbital Configuration

Prior to selecting the MEO orbital configuration, Inmarsat, the largest shareholder and sponsor of ICO, conducted an extensive analysis and review of a broad range of the then available technical options. These studies focused on service properties, cost, technical risks and market potential. The research was conducted by Inmarsat with Inmarsat Signatory participation, and through a number of commissioned studies by main aerospace and telecommunications industry contractors.

The Inmarsat studies considered the relative merits of the generally known technically feasible options of providing a communication service to handheld satellite phones from:

i) Low earth orbit (LEO – up to 2,000 km altitude)

ii) Medium earth orbit (MEO – 8,000 to 20,000 km) or

iii) Geostationary orbit (GEO)

To cover the earth fully, LEO requires around 40-70 satellites, MEO needs 6-20 satellites, and GEO needs 3-6 satellites.

Inmarsat concluded that the MEO configuration could offer best overall service quality for the desired market. This is because of the orbital properties, which confer, with a reasonable number of satellites, the following benefits:

i) High average elevation angle from user to satellites, minimising probability of blockage

ii) High probability of a user being in the field of view of more than one satellite hence offering good satellite path diversity

iii) Slow-moving satellites (about 1 degree per minute across the sky as perceived by the user)

The technology studies of the LEO, MEO, and GEO satellite constellations concluded that MEO represents a reasonable implementation and schedule compromise. The large number of LEO satellites which would be needed, taken with their relatively short lifetimes in their expected radiation environment, present logistical and manufacturing problems in maintaining the constellation.

Constellation Service Performance Features

There are trade-offs related to each of the orbital configurations with respect to service characteristics, costs, complexities and risks. Connectivity and propagation delay are most likely to be noticed by users.

Connectivity: All orbital configurations normally require a user to have a line of sight to a satellite to initiate and retain a connection. Obstructions such as buildings and mountains between a user and a satellite may prevent the establishment of a link or may sever an existing link, and both natural and urban canyons may present difficulties.

MEO or LEO systems typically allow for overlapping coverage of the earth from multiple satellites, which increases the likelihood of at least one satellite being within view of the user in order to establish a link. As the satellites in these systems move relative to the earth, they also leave the user's or the SAN's line of sight, moving either beyond the horizon or possibly behind an obstruction. To maintain a link, MEO and LEO systems must incorporate a system of "hand-offs".

MEO satellites move across the user's field of view more slowly than LEO satellites because they orbit at a higher altitude, which reduces the frequency of hand-offs required.

Propagation delay: This is the time delay, proportional to the sum of the distances between the satellite and the user and between the satellite and the SAN, resulting from the distance the radio signal travels. GEO systems have the highest propagation delay, while LEO systems encounter the least. A propagation delay of less than 200 milliseconds is characteristic of MEO and is well within acceptable limits.

Frequency plan

Service link spectrum requirements for connection between user terminals and satellites: The choice of bands for the provision of service links for MSS systems include 1.6/1.5 GHz, 1.6/2.4 GHz and around 2 GHz. ICO has chosen to operate in the 2 GHz bands, between 1985-2015 MHz and 2170-2200 MHz.

For the planned traffic distributions and volume, the service link spectrum requirement will be around 10 MHz in each of the uplink and downlink directions. The ICO satellites will retain the flexibility to maximise the use of the spectrum available in each direction in the 1985-2015 MHz and 2170-2200 MHz bands.

Feeder link spectrum requirements for connection between satellites and SAN): For feeder link operation, ICO has chosen to operate in the bands 5150-5250 MHz and 6975-7075 MHz. These bands form part of a pair of new allocations made by WRC-95 for feeder links to non-geostationary satellites providing MSS.

Ground segment

The satellites will be linked to a ground network (the ICONET) which will interconnect twelve SANs optimally located throughout the world. SANs comprise earth stations with multiple antennas for communicating with satellites, and associated switching equipment and databases. The ICONET and SANs will implement the selection of call routings to ensure the highest possible quality and availability of service to system users. Figure 3 shows a block diagram of the ICONET.

Gateways will be located throughout the world and will connect into the ICONET to serve as the interface with the PSTN and PLMNs. They will be owned by third party operators who will be authorised to access the ICO system.

ICONET and satellite access nodes

SANs will be the primary interface between the satellites and the Gateways. They will also house the equipment which will route the satellite signals for distribution to the appropriate Gateways. A SAN will comprise three main elements:

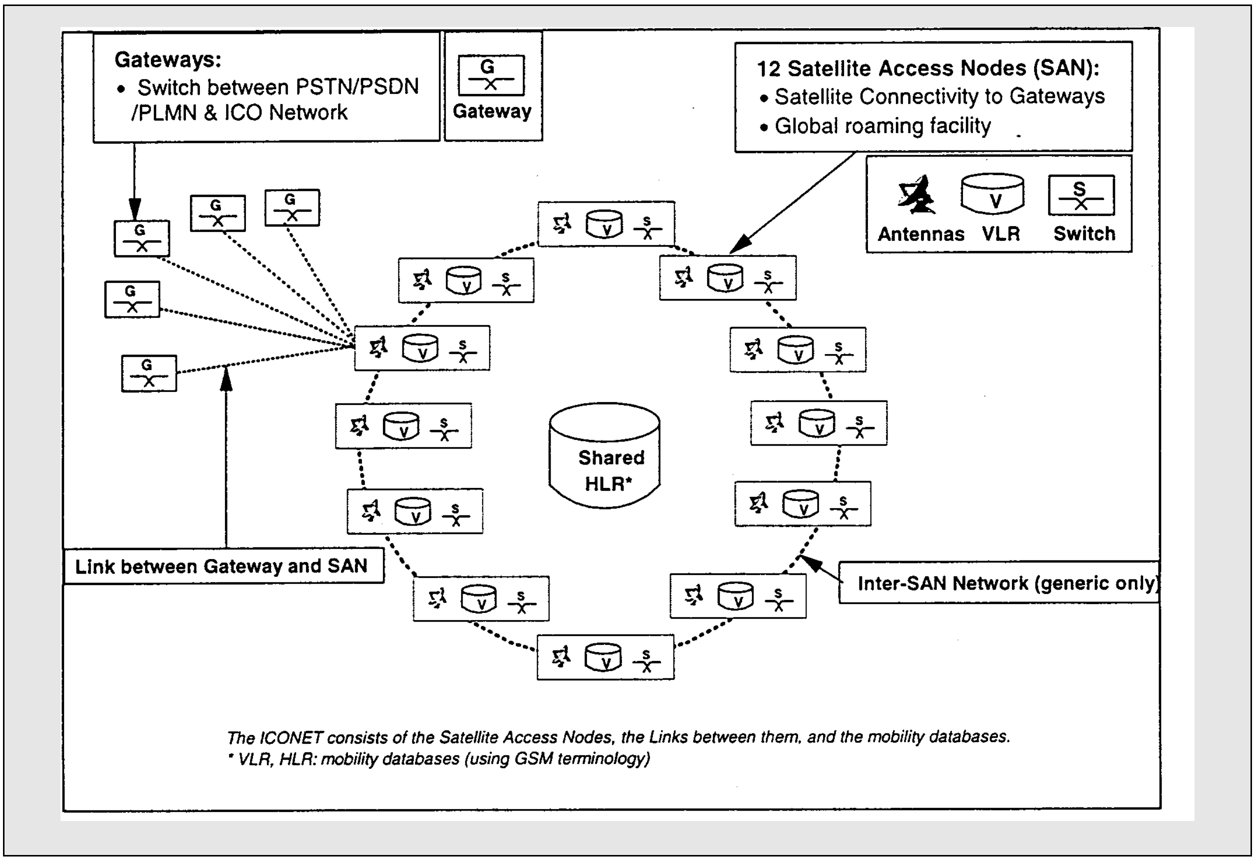
i) Five antennas, with associated equipment to communicate with the satellites;

ii) A switch to route traffic within the ICONET and to Gateways; and

iii) Databases to support mobility management.

Each SAN will contain a database to hold details of user terminals currently registered to that SAN (in GSM terminology, this is a Visitor Location Register or VLR). At least one SAN will also contain a database for holding some information on users (Home Location Register or HLR), though this function may be distributed among more than one SAN or in the PLMN.

Figure 3 – The ICONEt



Each SAN will track the satellites within its sight, direct communications traffic to the optimal satellite for the most robust link, and subsequently, as appropriate, will execute hand-offs from the satellite with the less robust link to the optimal satellite then in the line of sight of the SAN so as to maintain uninterrupted communication.

ICO will procure twelve SANs, strategically located around the world (approximately two per continent) and will own these facilities and commission their design and procurement, but the installation and operation of the SANs will be contracted out. The network of SANs will be interconnected by a terrestrial backbone network called the ICONET, which will allow calls to be routed through the ground segment of the network to the SAN best able to service the call. The ICONET will be managed by the Network Control Centre.

Gateways

Gateways are switches which will serve as the link between the SANs and the public terrestrial networks. Each Gateway will be connected to the ICONET, which will allow flexibility in traffic allocation and back-up traffic routing. The switch may be either a self-contained, dedicated unit or may be implemented as an additional function on an existing switch.

Gateways will be owned and operated by third parties. There is no pre-defined technical limit to the number of Gateways which can be incorporated into the system, and their location will be determined by market proximity and access to existing PSTN, PSDN, or PLMN switches.

User Mobility Management

In order to provide global roaming, the ICONET will include a system for management of global user mobility based upon the existing digital cellular standard, GSM.

HLRs in co-ordination with VLRs will verify user information and status, and locate the user anywhere in the world. Any handset which is turned on will send a signal via satellite and SAN to the user's HLR which will verify the user's status and allow access to the system. The system will communicate this clearance to that SAN and register it in its VLR. The HLRs second function is to communicate the VLR location of any user to the SAN through which an incoming call is originated. This will enable the call to be directed to the SAN closest to the intended call recipient. The call will then be completed via satellite link.

Cellular Network Integration

A critical feature of ICO will be its integration into PLMNs existing at that time. In most instances the satellite network will be viewed as a complementary service to PLMN subscribers who wish to have the capability of making and receiving calls in areas not serviced by their PLMNs.

The ICO network architecture will allow two broad groupings of users to be made, with corresponding service distinctions: local or regional users and users who operate worldwide.

ICO Global Communications is studying several alternative approaches to integration, including the system requirements for global numbering and the logistics of locating users and call linkage within and outside the range of cellular networks. ICO will include a dual-mode handset which will access both satellite and PLMN networks from the same terminal.

Telemetry, Tracking and Control – TT&C

TT&C stations will manage the ICO satellite system by tracking the movements of the satellites and adjusting their orbits to maintain the constellation. TT&C stations will also monitor the general condition of the satellites by collecting data on the power supply, temperature, stability and other operating characteristics of the satellites and relaying that data to the stations for processing and response.

Satellite Network Control

The NCSs, acting through TT&C stations and SANs, will control the transponder linkages between the feeder and service antennas on the satellites. This process will dictate, among other things, frequency reconfiguration within feeder link beams and optimal channel allocation between high and low traffic spot beams.

User terminals

Pocket Phones

The large majority of ICO user terminals are expected to be handheld, pocket-sized telephones, capable of dual-mode (satellite and cellular or PCS) operation and very similar in size and appearance and in voice quality to today's pocket cellular/PCS phones. The price of ICO dual mode phones, on the basis of high volume production, is expected to be competitive with other comparable satellite systems at service introduction.

The ICO pocket phone is planned to have optional features including external data ports and internal buffer memory to support data communication, messaging functions, facsimile, and the use of smartcards (SIMs).

Safety

The ICO system has been designed to ensure that the pocket phone will comply with expected safety requirements in respect of radio frequency radiation. The average transmitted power typically during use will not exceed 0.25 watts. Existing cellular phones have average transmitted power typically in the range of 0.25 to 0.6 watts.

Other Derived User Terminal Types

The technology used in ICO pocket phones is expected also to be incorporated in a wide range of other user terminal types including vehicular, aeronautical, and maritime mobile terminals and semi-fixed and fixed terminals, such as rural phone booths and community telephones. For many of these terminals it will be possible to use higher-gain antennas and/or higher transmit powers than are used for the pocket phones, which will permit higher bit-rate services to be supported.

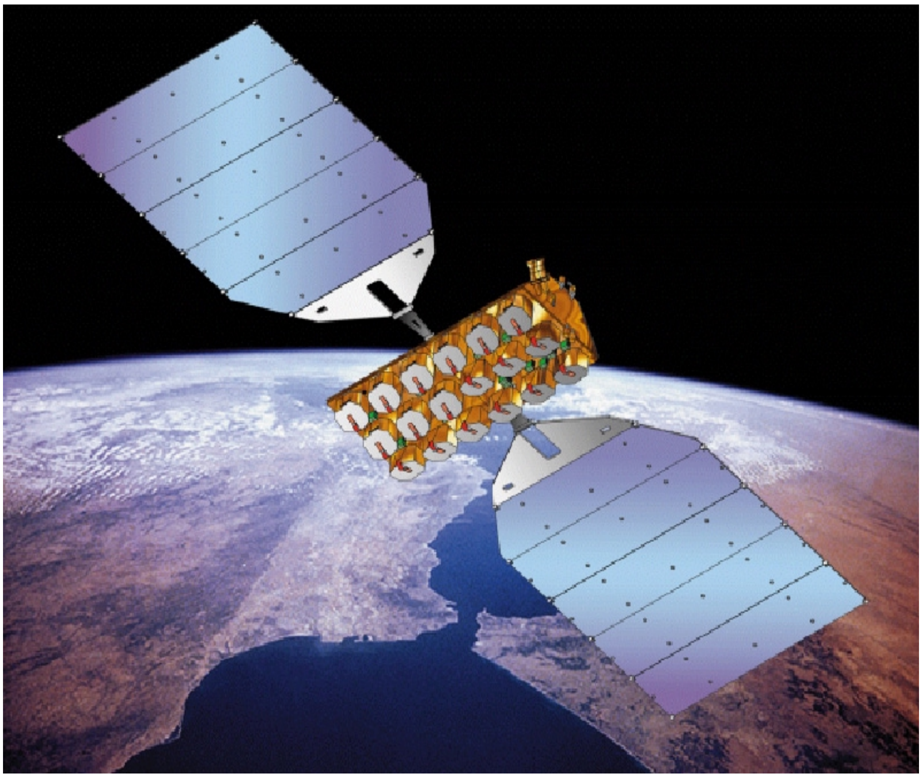
ANNEX 2E

The skybridge: global multimedia access

The following information is based on information provided by Skybridge:

Introduction

Over the past few years, businesses and individuals have rapidly embraced the Internet, realizing the benefits not only of being able to access information from all over the world at low cost, but also the potential of electronic commerce. However, the powerful interactive multimedia services that characterize the new information age required ever larger circuit capacities to transport text, images, video, sound and data. Indeed, many of these services require transmission rates tens to thousands of times larger than those needed for a simple phone call. It is estimated that by the year 2005, more than 400 million users worldwide will be using such broadband services.



To meet this demand, telecom operators have invested heavily in fiber-optic backbone networks which can transport multimedia information around the globe at the speed of light. However, the local networks that bring these services to end users have lagged far behind, creating bottlenecks which will be familiar to everyone who has waited and waited for an Internet page to download. The main reason is, of course, the high cost of replacing or upgrading the local access infrastructure, which represents more than 70% of total network investment. It is both financially and physically impossible to replace the installed local loop by high speed fiber within a few years.

Technologies such as Asymmetric Digital Subscriber Line (ADSL) and cable modems are providing attractive solutions that use the existing copper telephone networks and cable networks. However, such technologies cannot be made available technically or at a sufficiently low cost in a number of situations (e.g. in low density areas, where existing networks are of insufficient quality). Many users will thus continue to suffer from traffic jams. As a result, operators are losing the considerable revenue which they might expect to earn from offering broadband services to all their subscribers.

The question then is, how can we solve this problem in an affordable way? The answer is SkyBridge, a satellite access system which overcomes the local loop limitations by providing high speed access to the world's fiber optic backbone networks. Scheduled to start operation in 2003, SkyBridge will enable telecom operators to offer local broadband access to over 20 million users worldwide, using a constellation of 80 Low Earth Orbit (LEO) satellites. It will be the first broadband LEO satellite system, and the only one focused entirely on local access. By interconnecting the SkyBridge satellite access network with fiber optic backbones, SkyBridge will make global high speed end-to-end connectivity a reality at an affordable price. Operators will benefit from low cost, flexibility, rapid deployment and a high quality of service. As SkyBridge provides "bandwidth on demand", they can allocate capacity wherever and whenever necessary to meet users' needs. Users gain the advantage of cost-effective access to multimedia services using a small, inexpensive antenna.

Because it uses satellite technology, SkyBridge is not constrained by the terrain or the existing local infrastructure. Thus it enables people and businesses in rural and remote areas, and in developing countries, to take advantage of the information age, helping to boost the local economy.

SkyBridge System

SkyBridge uses a constellation of 80 satellites orbiting the earth at an altitude of 1469 km. Because the system operates in the Ku-band (10 to 18 GHz), it is able to provide high availability based on the use of proven satellite technology. In addition, the Ku-band is much more robust against rain attenuation than higher frequency bands, and therefore offers a high quality service even in poor weather conditions. The use of this band, which SkyBridge shares with a number of geostationary satellite systems and terrestrial microwave systems, has been unanimously approved by the World Radiocommunications Conference (WRC) of the International Telecommunications Union (ITU).

Traffic is routed from the user's antenna via the LEO satellites to a terrestrial gateway which interfaces with the terrestrial infrastructure. Switching and routing can be performed either in the gateway or by a remote switch in the operator's network.

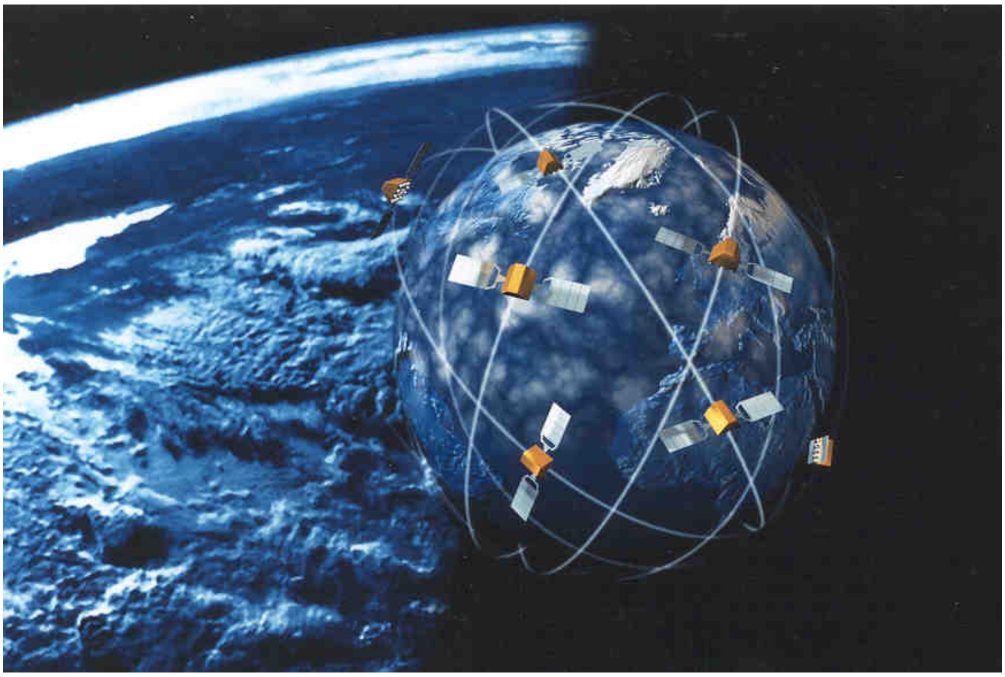
Flexibility and the ability to evolve gracefully are built into SkyBridge as it operates on the simple "bent pipe" principle with no on-board switching or satellite-to-satellite communication links. Consequently, operators can manage the capacity of their networks as necessary, and develop services to meet emerging user demands. This flexibility will be crucial in maintaining a competitive market position well into the next millennium.

Architecture

The system architecture is divided into a space segment and a telecommunication segment. The space segment comprises:

- A constellation of 80 LEO satellites, plus spares.
- Two Satellite Control Centres (SCC).
- Tracking, Telemetry and Command (TT&C) ground stations.
- Two Mission Control Centres.

The satellites orbit in a so-called Walker constellation, which consists of 20 planes, equally inclined by 53º relative to the equator. There are four satellites in each plane, orbiting at an altitude of 1469 km.



A key design feature of the satellites is the use of active antennas which generate the spot beams and maintain them pointing towards the corresponding terrestrial gateways. Each spot beam illuminates a 700 km diameter cell. It can serve user terminals with an elevation angle of more than 10°.

The above figure shows the telecommunication segment. This consists of user terminals and gateways, which interface with local servers, narrow- or broadband-terrestrial networks, or leased lines. User terminals comprise the antenna equipment and an interface for connection to the user's equipment, such as a multimedia PC. Up to 200 gateways are planned, ensuring global coverage. In regions with low population densities, one gateway can serve several cells.

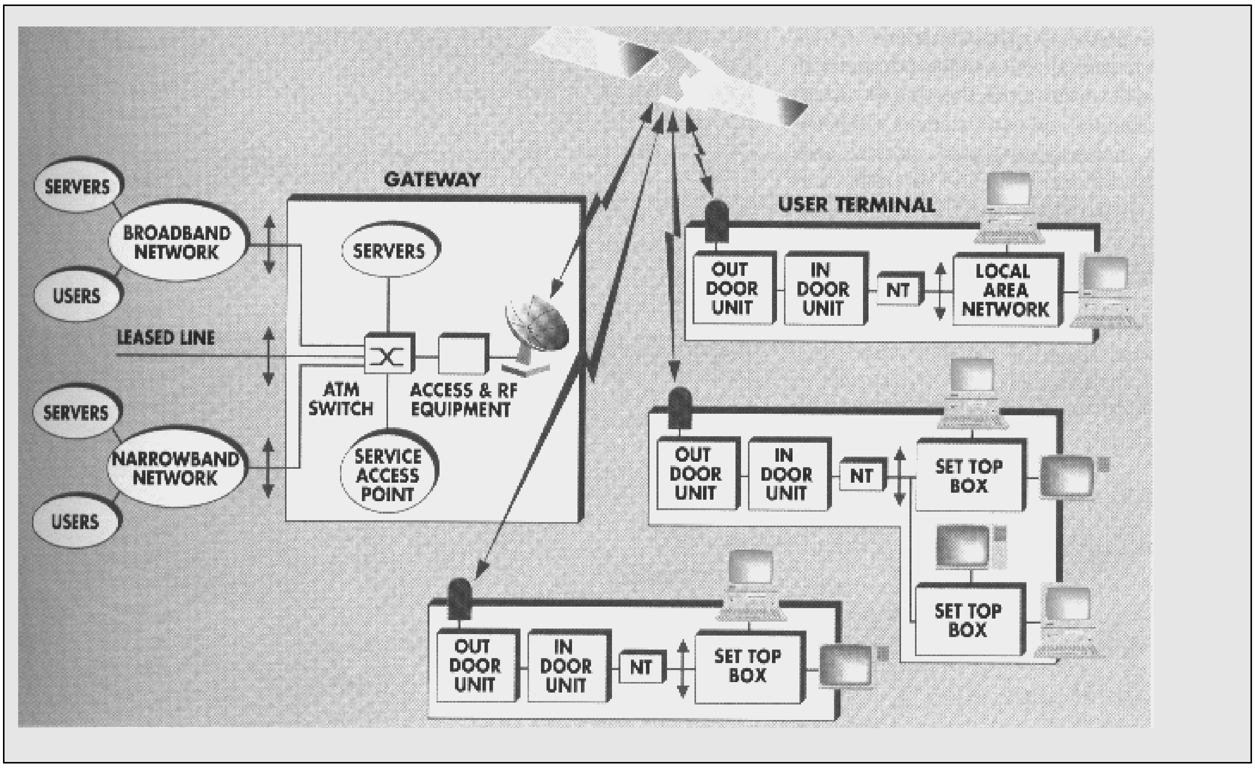
Residential users equipped with small, 50 cm roof-mounted antennas can receive at up to 20 Mbit/s (forward link) and transmit at up to 2 Mbit/s (return link). Business terminals using 80 to 100 cm antennas can receive and transmit 3 to 5 times higher bit-rates.

SkyBridge uses standard Asynchronous Transfer Mode (ATM) and Internet Protocol/Transmission Control Protocol (IP/TCP) so that it can work seamlessly with the established and future terrestrial networks.

Handover

As a satellite moves out of sight of a terminal, the traffic it is handling with that terminal is handed over to another satellite to ensure continuity of service; the first satellite then refocuses on other terminals. Handover for a cell is managed by the corresponding gateway.

Figure 1 – Telecommunication segment of the SkyBridge system



Links with the outgoing satellite are maintained long enough for the terminal and gateway antennas to point to and synchronize with the incoming satellite. Calls are switched over as soon as synchronization has been achieved.

SkyBridge Services

Because of the constraint of data acknowledgement, the performance (throughput, window size) of IP packet transport is mainly influenced by the satellite round trip delay. The low signal transmission delay of the SkyBridge constellation (30 ms compared to 500 ms for Geostationary satellites) guarantees a good transfer efficiency for the support of Internet access services.

In addition, SkyBridge is designed to be compatible with current and emerging Internet protocol standards (e.g. Resource Reservation Protocol/Differentiated Services, Point to Point Protocol over Ethernet) which are currently being defined to support real-time quality of service over the Internet or other packet-based broadband networks.

End-user services

The SkyBridge System supports a wide range of services :

- Multimedia Applications Over the Internet: meet the increasing demand for real-time media (e.g. live video press conferences, live Internet radio), as well as emerging telephony and video conferencing on the Internet.
- Direct access to on-line services and content: SkyBridge System traffic can be routed directly to a local server that provides local services and content.
- LAN interconnection and private networking: these are provided by interconnecting individual terminals and Local Area Networks (LANs) via satellite links. In addition, the System enables operators to provide virtual LAN services by facilitating remote access to LANs (for telecommuters at home or employees in the field), and Wide Area Network (WAN).
- Connection to the Public Narrowband Network: enables any end user to use his or her own standard or digital telephone set, as well as other types of End User Terminals, such as screen phones or set-top boxes.
- Videotelephony and Video Conferencing: enable multiple users at two or more sites to communicate in real-time with bi-directional or multi-directional exchange of audio, video, and other data (including corporate applications and documents).
- Electronic Commerce: includes the marketing, ordering, delivery and payment for goods and services online. E-commerce is expected to be a major driver force behind the demand for broadband services over the Internet, intranets and extranets.
- Telecommuting: makes it possible for people to work away from traditional offices through remote LAN/WAN connections. The SkyBridge System also gives telecommuters access to remote LAN connections.
- Telelearning: supports "virtual classroom" in which instructors and students are in different locations served by SkyBridge.
- Telemedicine: involves a combination of video-on-demand, multimedia information retrieval and video conferencing.

Narrowband and Broadband transport services

SkyBridge supports narrowband services both for the provision of telephony services in suburban areas and sparsely populated areas, and for the rapid deployment of public infrastructures and private networks.

In addition SkyBridge's transport capability supports connection to complementary broadband access systems (such as ADSL), and the interconnection of remote groups of users to the SkyBridge Gateways.

SkyBridge end-users

The SkyBridge system serves two categories of users:

- professional users accessing remote databases and sharing interactive applications with other parties outside their business Customer Premises Network.
- residential users requiring entertainment, video communication, access to Internet Service Providers, etc.

Professional users

Professional users may require point to point or point to multipoint connectivity for applications such as teleconferencing, cooperative work, LAN-to-LAN interconnection.

Via SkyBridge, a professional user can contact another professional user connected to an external corporate network (e.g. WAN, Intranet, Virtual Private Network (VPN),) or any other user equipped with a SkyBridge terminal.

Residential users

Communications can be set up between different local SkyBridge residential users within the SkyBridge coverage area as well as between any residential user and the Service Providers connected to SkyBridge gateways.

Operators

In certain cases, network operators can also be considered as end users. In particular, SkyBridge can transmit switched or transparent E1 and T1 links.

Frequency Reuse Concept

One of the most important aspects of efficient frequency management is the concept of sharing between different telecom services and systems. Well-structured sharing techniques, enable new technologies to be deployed, which allow exciting new services to come on-line without disturbing existing services. SkyBridge is an example of this spirit of sharing. Innovative design enables SkyBridge to greatly extend access to advanced telecommunications by re-using frequencies already used by other systems, including geostationary satellite and fixed service systems.

SkyBridge is playing a pivotal role in creating the environment, that will allow this tradition of sharing and efficient frequency use to continue for Non-Geostationary Satellite Orbit Fixed Satellite Service (NGSO FSS) systems. The 1997 World Radiocommunication Conference (WRC-97) took a strategic first step in approving the concept of Power Flux Density (PFD) limits. These specify the technical parameters with which a constellation of NGSO satellites must comply to ensure adequate protection to geostationary satellites and terrestrial services. WRC-97 also created a technical forum in which engineers from around the world are meeting to determine at the precise levels of protection to be afforded to Geostationary Satellite Orbit (GSO) systems and terrestrial services, while allowing the introduction of NGSO FSS services. This cooperative effort is well underway and has made substantial progress towards its goal. It will be up to WRC-2000 to finalize this regime and adopt the work done by the ITU-R working group examining this subject.

Because SkyBridge shares the Ku-band (10 to 18 GHz) with existing geostationary broadcasting satellite systems, terrestrial networks, space science projects and radiolocation services, one of the main design challenges was to ensure that it will not cause harmful interference to these or any future systems using the same band.

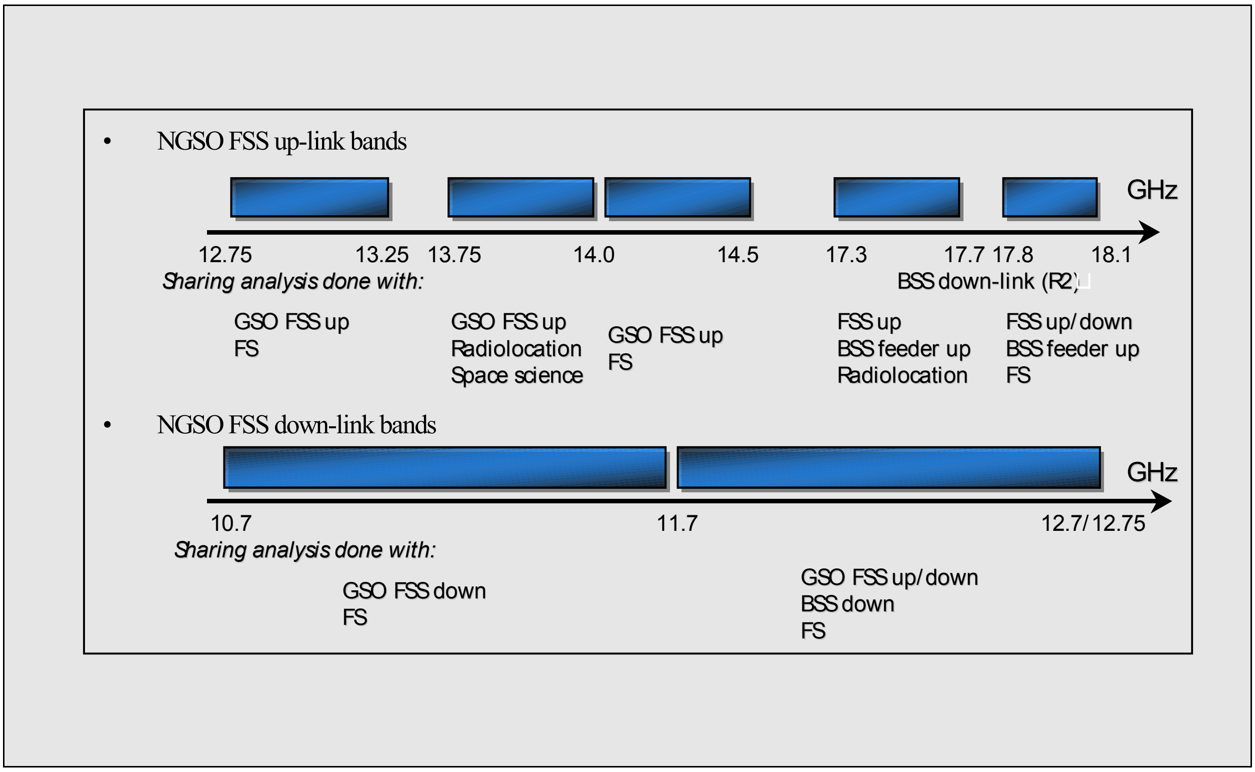
To achieve this, SkyBridge has optimized the various system radio parameters. First, the frequency plan has been carefully selected, including the allocation of some bands exclusively to gateway stations with no user terminals. Second, the satellite antennas have been designed to provide optimum performance. And third, the waveform has been carefully designed to minimize the required power and to ensure that the system is robust to interference from other users of the Ku-band.

Protecting Geostationary Satellite Systems

Existing geostationary satellite systems operate within a well established operating environment. Frequency sharing with geostationary satellite systems using the same Ku-band is based firstly on giving adequate protection against three potential forms of interference:

- Interference between the main beam of a SkyBridge transmitter and the main beam of a geostationary satellite receiver.
- Interference between the main beam of a SkyBridge transmitter and the sidelobe of a geostationary satellite receiver.
- Interference between the sidelobe of a SkyBridge transmitter and the sidelobe of a geostationary satellite receiver.

Figure 2 – Frequency plan and sharing issues



By exploiting the directivity of the SkyBridge and geostationary satellite earth stations, SkyBridge protects existing satellite systems against these forms of interference in a way that is transparent to end-users. All GSO earth stations point toward the GSO arc, offering the rest of the space an important antenna angular discrimination. In particular, to avoid interfering with the receivers of geostationary systems, SkyBridge satellites cease transmission to a given cell when an earth station recognizes that they are close to a GSO pointing direction.

Each gateway has a defined "non-operating zone" which includes all the satellite positions that would potentially create interference with a geostationary satellite and associated earth stations. As soon as a SkyBridge satellite enters this zone, it ceases transmission to the gateway cell corresponding to the exclusion zone; at the same time, the gateway and all the user terminals in that cell stop transmitting to the satellite. Traffic in the cell is transparently handed over to other satellites in the constellation to ensure continuity of service.

Based on an extensive analysis of the interference environment, the non-operating zone has been defined as a belt of ±10º on either side of the geostationary arc, as seen by any earth station in a gateway cell. Consequently, the non-operating zone is wider than ±10º for most points within the zone, ensuring even better protection. This limits the residual main-beam-to-side-lobe and side-lobe-to-main-beam power.

Protecting Terrestrial Services

The primary measure taken by SkyBridge to help protect the fixed terrestrial service against interference was to carefully select the frequency bands used by the small user terminals when receiving and transmitting. Only gateways operate in frequency bands that are regularly used by the fixed service. Coordination between the two systems is eased as a result of the better discrimination provided by the large gateway antennas and the limited number of earth stations to be coordinated. While this constrains the SkyBridge system, it simplifies the protection of fixed service links operating in the Ku-band.

Careful refinement of the propagating models used in coordination procedures has minimized the distance between an earth station and a fixed service link below which the two parties involved need to coordinate their operations. General agreement has been reached on these procedures.

Fixed Service reception also needs to be protected from NGSO satellite emissions through the definition of adequate PFD limits. General agreement has also been reached on these limits.

Protection of Radiolocation Service and Space Science

Existing radio regulations stipulate a maximum Equivalent Isotropic Radiated Power (EIRP) emitted by earth stations in order to protect Space Science systems which are planned to operate in the Ku-band. They also specify the minimum EIRP at which earth stations must operate in order to be protected from the radiolocation service.

SkyBridge has undertaken a detailed technical analysis to determine how it will affect these services. As a result of its use of an interference robust waveform SkyBridge has proven that it can operate in the Ku-band alongside radiolocation services.

In addition, SkyBridge is proposing to limit the power transmitted by its gateways below the level defined in current radio regulations, thereby ensuring that space science applications are protected against interference from NGSO earth stations.

ANNEX 2F

The teledesic system

The following information is based on information provided by Teledesic:

Through a broad, cooperative effort, Teledesic will bring affordable access to fiber-like telecommunications services to all parts of the world that would not be economical to serve through terrestrial means.

Today, advanced telecommunications infrastructure is limited to the developed urban areas of the world. This leaves most of the world's population without access to even basic communications services. Even those areas with basic voice service get access through 100-year-old technology – analog, copper networks – that for the overwhelming part will never be upgraded to support digital, broadband services.

Teledesic is building a global, broadband "Internet-in-the-Sky." Using a constellation of several hundred low-Earth-orbit satellites, Teledesic will create the world's first network to provide affordable, worldwide, "fiber-like" access to telecommunications services such as broadband Internet access, videoconferencing, high-quality voice and other digital data needs. On Day One of service, Teledesic will enable broadband telecommunications access for businesses, schools and individuals everywhere on the planet.

Teledesic has received support from the developed and developing world alike, resulting in both international and domestic satellite service designations for the frequencies necessary to accommodate the Teledesic Network. In March 1997, the U.S. Federal Communications Commission licensed Teledesic to build, launch, and operate the Teledesic Network.

The Teledesic Network

Teledesic does not intend to market services directly to end-users. Rather, it will provide an open network for the delivery of such services by others. The Teledesic Network will enable service providers in host countries to extend their networks, both in terms of geographic scope and in the kinds of services they can offer. Ground-based gateways will enable service providers to offer seamless links to other wireline and wireless networks, such as the Internet.

The Teledesic Network will consist of 288 operational satellites, divided into 12 planes, each with 24 satellites. To make efficient use of the radio spectrum, frequencies are allocated dynamically and reused many times within each satellite footprint. Within any circular area of 100 km radius, the Teledesic Network can support over 500 megabits per second (Mbit/s) of data to and from user terminals. The Teledesic Network supports bandwidth-on-demand, allowing a user to request and release capacity as needed. This enables users to pay only for the capacity they actually use, and for the Network to support a much higher number of users.

Teledesic will operate in a portion of the high-frequency Ku-band (28.6-29.1 GHz uplink and 18.8-19.3 GHz downlink). The Teledesic Network's low orbit eliminates the long signal delay experienced in communications through traditional geostationary satellites and enables the use of small, low-power terminals and antennas, about the size of direct broadcast satellite (DBS) dishes.

The Teledesic Network is designed to support millions of simultaneous users. Most users will have two-way connections that provide up to 64 Mbit/s on the downlink and up to 2 Mbit/s on the uplink. Broadband terminals will offer 64 Mbit/s of two-way capacity. This represents access speeds up to 2,000 times faster than today's standard analogue modems. For example, transmitting a set of x-rays may take four hours over one of today's standard modems. The same images can be sent over the Teledesic Network in seven seconds.

Design, production and deployment of the Teledesic Network will cost $9 billion. End-user rates will be set by service providers, but Teledesic expects rates to be comparable to those of future urban wireline services for broadband access.

Seamless Compatibility with Terrestrial Networks

Without knowing for certain all the applications and data protocols a broadband network will be called upon to accommodate in the 21st Century, it is reasonable to assume that those applications will be developed in the advanced urban areas of the developed world – where fiber-optics sets the standard. Satellite systems offer the capability to provide location-insensitive, switched, broadband access, extending the reach of networks and applications to anywhere on Earth. But to ensure seamless compatibility with those networks, a satellite system should be designed with the same essential characteristics as fiber networks – broadband channels, low error rates and low delays.

Satellite systems are of two general types: geostationary-Earth-orbit (GEO) and non-geostationary, primarily low-Earth-orbit (LEO). Geostationary satellites orbit at an altitude of 36,000 kilometers (km) above the Equator – the only orbit that allows the satellite to maintain a fixed position in relation to Earth. At this height, communications through a GEO entail a minimum round-trip transmission latency – end-to-end delay – of at least one-half second. This means that GEOs can never provide fiber-like delays.

This GEO latency is the source of the annoying delay in many intercontinental phone calls, impeding understanding and distorting the personal nuances of speech. What can be an inconvenience on voice transmissions, however, can be untenable for real-time applications such as videoconferencing as well as many standard data protocols – even for the protocols underlying the Internet.

One of the fundamental principles of the Internet is the notion of all applications moving on to a common network platform – an open network based on common standards and protocols. The idea of stand-alone, proprietary networks, or application-specific networks, is fast disappearing. All applications will move over the same networks, using the same protocols. In these packet-switched networks – where voice, video, and data are all just packets of digitized bits – it is not practical to separate out applications that can tolerate delay from those that cannot. As a result, the network should be designed for the most demanding application. The Teledesic Network is designed to provide end-to-end Quality-of-Service that enables global enterprise networking, meeting the demands of the Internet of the future.

Distributed vs. Centralized Architecture

Just as networks on the ground have evolved from centralized systems built around a single mainframe computer to distributed networks of interconnected PCs, space-based satellite networks are evolving from centralized networks relying on a single geostationary satellite to distributed networks of interconnected low-Earth-orbit satellites. In geostationary systems, any single satellite loss or failure is catastrophic to the system. To reduce this contingency to acceptable levels, reliability must be engineered far along toward the point of diminishing returns where further gains in reliability are achieved only at a very high cost.

With a distributed network, like the Teledesic Network, reliability can be built into the network rather than the individual unit, reducing the complexity and cost of the individual satellites and enabling more streamlined, automated manufacturing processes and associated design enhancements. In its distributed architecture, dynamic routing, and robust scalability, the Teledesic Network emulates the most famous distributed network, the Internet, while adding the benefits of real-time capability and location-insensitive access.

Low-Earth-Orbit Satellite Systems

The evolution from geostationary to low-Earth-orbit (LEO) satellites has resulted in a number of proposed global satellite systems, which can be grouped into three distinct types. These LEO systems can best be distinguished by reference to their terrestrial counterparts: paging, cellular, and fiber.

The Big LEOs, for example, provide premium-priced, narrowband mobile voice service, whereas Teledesic provides primarily fixed, broadband connections at costs comparable to urban wireline service. Just as cellular and fiber are generally not considered to be competitive, the only thing Teledesic really has in common with the Big LEOs is the use of low-Earth-orbit satellites.

Elevation Angle

The Teledesic Network is designed so that from anywhere on Earth, a Teledesic satellite can always be viewed nearly directly overhead. This is ensured by having an elevation angle of 40 degrees or higher at all times in all locations.

Teledesic's 40° elevation angle enables users to place terminals on most offices, schools, and homes with an unobstructed view of the sky in all directions. A lower elevation angle dramatically increases the likelihood of obstruction by surrounding buildings, trees, or terrain, preventing service. In many areas, a low elevation angle can make any service impractical or simply impossible.

Additionally, signals at high frequencies can also be blocked by rain, especially when sent at a lower elevation angle. Teledesic's 40° elevation angle is essential to meeting the company's goals for high quality-of-service, with availability comparable to terrestrial networks. It also reduces the user terminal size and cost and improves the ease of coordinating the use of radio frequencies with other systems and services.

The Market for Teledesic

The convergence of computing and communications is causing all things one associates with a high standard of living – from education and health care to economic development and public services – to become dependent on an ever-increasing flow of information. In highly urbanized areas, this demand for information is being satisfied by the high-bandwidth and high-quality connections of fiber optics. Increasingly, institutions and individuals are utilizing broadband connections for Internet access, computer networking, aggregation and trunking of voice lines, and telecommuting. But step out of the cities, and these fiber-like telecommunications services become prohibitively expensive or are simply unavailable at any price.

The Teledesic Network will seamlessly extend the existing terrestrial, fiber-based infrastructure to provide advanced information services anywhere on Earth. Customers will range from the information workers unwilling to be confined in increasingly congested cities, to countries backhauling aggregated voice lines from remote cellular sites, to multinational corporations connecting branch offices throughout the world into their existing global enterprise networks. Whenever and wherever institutions and individuals want access to the fiber-like telecommunications services currently available only in the most highly developed urban areas, the Teledesic Network can provide seamless connectivity.

And, because Teledesic satellites move in relation to the Earth, Teledesic provides the same quality and capacity of service to all parts of the globe. In this sense, Teledesic's Internet-in-the-Sky is an inherently egalitarian technology. On Day One of service, Teledesic will help transform the economics of telecommunications to enable universal access to the Internet and the information age.

ANNEX 2G

The final analysis system

The following information is based on information provided by Final Analysis:

One of the most aggressively pursued investment, service and manufacturing opportunities in the world today resides in the newly emerging GMPCS industry. GMPCS is the acronym for global mobile personal communications by satellite – space-based telecommunication infrastructures for the next millennium.

Industry analysts and leading investors predict a booming market for new communication services delivered by global satellite constellations. These communications systems range from tens to hundreds of small satellites circling the earth and synchronized for seamless service anywhere. This is a new investment field, rapidly being filled. Only a few early investment opportunities remain as the various players rush to complete their financing, implement their systems, and begin offering customer services.

Within the GMPCS industry, there are distant differences in system technologies, costs, and target markets. Some satellite systems are designed to provide voice services, catering to business travellers using hand-held mobile phones. Other satellite systems are geared to offer sophisticated computer-to-computer data networks in the sky.

There is yet another group, Little LEO systems, capturing the imagination of investors, business entrepreneurs and markets. Because of the unique high-margin market opportunities, tested and reliable technology, and service affordability of Little Leo systems, Final Analysis Communication Services Inc. (“Final Analysis” or “FACS”) has chosen to operate in this segment of the GMPCS industry.

Final Analysis is already positioned to be the low-cost provider and one of two industry leaders. Far from being a “paper concept” company, Final Analysis has designed, built and launched two satellites; built, tested, and implemented three fully operational ground stations; built, tested and implemented a satellite control center; secured launch services for the satellite constellation; and built working prototype user terminals. Digital data applications and market development are underway in several key market segments. National and international service development at a cost of over four times the Final Analysis capital investment.

Little LEO satellite systems are focused on affordable messaging services targeted to the world’s underserved markets and on low-cost two-way messaging, asset tracking and data monitoring applications. These new services will be available in the near term with high volumes and low costs to target populations and industries in industrialized and developing countries alike. An advantage for Little LEO services is their ability to cost-effectively serve less developed and remote regions at a profitable margin.

The term, “Little LEO”, was originally conceived to distinguish these *digital data* systems from so-called “Big LEO” *voice* systems. Both will launch their satellite constellations into low-earth orbit, hence the word LEO. Little LEOs operate in the UHF/VHF band below 1 GHz where the radio communication technology has been developed, tested, and successfully used since the 1970s. Big LEO systems will require significant research and development to produce radio transceivers that can reliably operate at higher frequencies at a reasonable cost, thus representing a greater technology risk than in the Little LEO market.

Already, Final Analysis has an initial focus on a strong portfolio of high-demand and high-margin services including:

Personal Messaging and Voice Mail: The Final Analysis satellite system will provide two-way paging and variable messaging (alphanumeric keypad and display) with message sizes exceeding current paging systems. Final Analysis voice messaging (microphone and voice digitizer/decoder) is an ideal solution for people unable to read or write as the service relays the recorded speech after it has been digitized into a data transmission. Final Analysis voice messaging provides a low-cost, instantly available communications system for rural areas and small villages in the developing countries that similarly lack power and phone service. Flexible Final Analysis satellites can also send large computer and data file.

Tracking and Asset Management: Final Analysis ground terminals (equipped with GPS receivers) will provide quick, accurate, low-cost positioning information for cargo, shipping containers, rail cars, barges and trucks. Control information can be easily provided through the same equipment. Cars can have a diagnostic "check up" monitored and reported by satellite. Anti-theft devices can report on the location of a stolen vehicle anywhere in the world and disable the vehicle through a satellite command. Commercial trucks can have customs control handled through remote sensors, which can verify that the container was not opened. Anything that moves can be tracked, thus protected.

Data Monitoring and Control: Final Analysis ground terminals (equipped with microprocessors) will provide memory and operational systems to monitor consumable supplies in vending and copier machines, climate conditions at weather stations, crop conditions and feed supplies on large farms, utility usage at homes and businesses, water quality conditions from ocean buoys, point-of-sale reports from retail stores and a wide variety of inventory management data. Currently some companies, such as utilities, are trying to implement wireless systems, but these efforts still require ground-based personnel to collect the data from individual locations by driving by the site. Satellites can read many tens of thousands of sites in one pass, providing far greater efficiency and lower cost.

CHAPTER 3

3 Digital switching systems

3.1 Introduction

3.1.1 Concepts

The fundamental task of telecommunications is to transfer messages. The communication system must ensure that the messages arrive at the correct receiver. The message transfer consists of the conversion of a message into signal units, the transport of these signal units, and the reconstruction of the message from these signal units.

Strictly speaking, the message transfer consists of switching as well as transmission. The transmission technology makes channels available for information transmission for long periods of time. But even this availability though, is flexible and can be varied. In the early days of transmission technology, flexibility was guaranteed by the distribution frame: Nowadays management commands are used to establish and direct transmission pathways. Following the further development of the control systems, transmission systems have begun to develop characteristics that have become more and more similar to those of switching technology. The major remaining difference is the control system, which uses measures of the network management (transmission technology) or signalling during connection set-up (switching technology). Both technologies are rapidly converging.

Switching network

The connection of terminal equipment, between which messages are to be exchanged, is performed by a switching network.

The switching network must be able to perform the following basic tasks:

- At any time, from every piece of terminal equipment or from every entry point, a connection to all terminal equipment on the network or the transfer to other networks must be possible in principle.

- Every connection must be controllable by the user.

On the one hand, the network must be in the position to fulfil the expected connection requests with sufficiently high probability, and to satisfy guaranteed quality parameters.

The technical effort to satisfy connection requests must, on the other hand, be reasonably limited.

The switching network is structured according to different points of view:

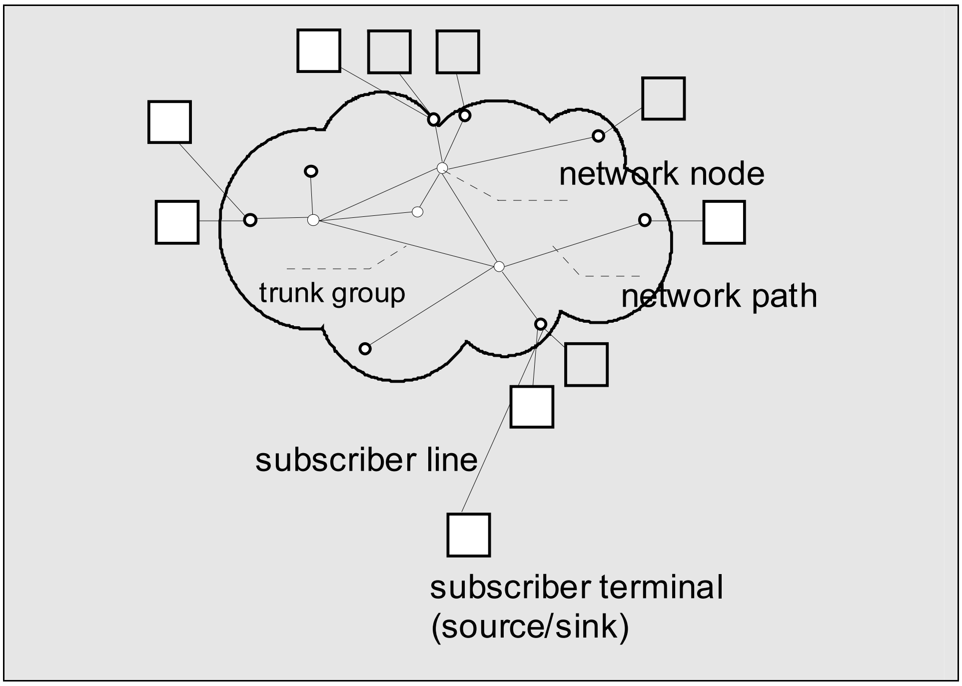
- requirements of the switching principle employed,

- amount of traffic,

- technical and economic parameters of the technology utilised,

- regulatory requirements.

Figure 3.1 – Switching network



The most important elements of the network are the nodes and paths. The payload between the network nodes is transported in the paths. Network edges are connection lines which link the terminal equipment on the network and are connection trunks between the network nodes and users. Groups of connections or channels between these same network nodes are brought together in trunk groups. The payload is determined in the network nodes.

Connections

A connection is a coupling of at least two pieces of terminal equipment on network access interfaces, network paths and network nodes of a network for the purpose of exchanging information.

For all forms of information exchange the rule is: at first, a connection through the network must be created. This connection can exist continuously or it can be created for a certain time period. If the connection has been created for a limited period of time, then there must be switching. A connection then exists for the duration of the complete information transmission (for example, in a telephone network) or the time for the transmission of a part of the information (for example, in ATM networks). The switching is carried out in the network nodes.

A switching process is always carried out in connection with a definite communication relationship.

Switching

Switching is the creation of connections for a limited period of time in a network by means of connecting channels, which make up the partial segments of the connection. Switching is the creation of the connection by means of control signalling.

Switching technology

All technical equipment which is used for the switching in a network can be designated switching technology.

The switching technology ensures that the information in a network, according to the switching principles current in this network, reach exactly those network nodes or subscribers for which they were designated.

From the point of view of the user of a network, switching is a service that can be employed in order to exchange information with one or many other users on the network.

A switching node is that part of a network where by evaluating technical switching information, partial segments of the network are put together for a connection. Simultaneously, depending on the traffic volume, the traffic of many terminals on the network is concentrated on a few paths of the network by switching.

The place where a switching node is located is called an exchange.

Switching nodes are distinguished according to their location in the network hierarchy as well as well as by their technical configuration.

3.1.2 Switching Principles

The switching principle is the way the switching of connections or messages is carried out.

Connectionless transmission

The connectionless mode is appropriate for networks in which sporadic, short information segments must be exchanged between the terminals, such that the time required for setting up and terminating a connection can be reduced. For this reason, these networks have mainly developed for communication between computers. The disadvantage of this kind of network is that all nodes are loaded with traffic, even if the information is not intended for them.

Connection-oriented transmission

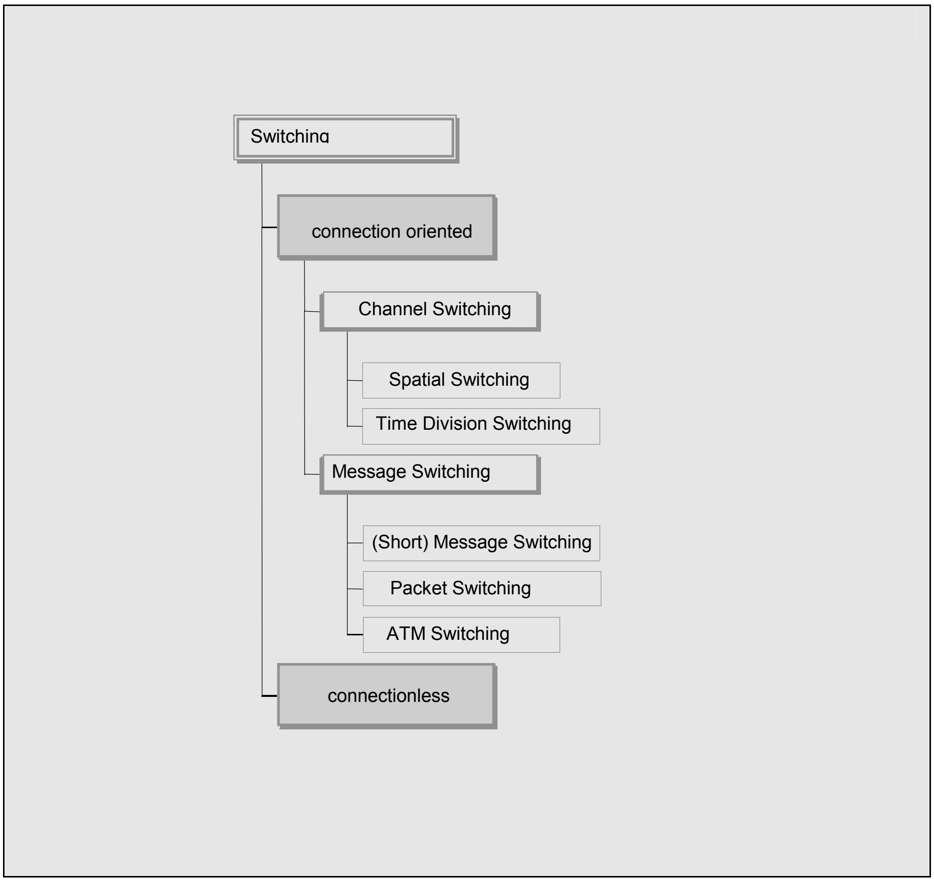
If the time required for the set-up of a connection is short compared with the time period that the connection exists, then connection-oriented service modes are more advantageous. Information is transported only to nodes that are necessarily involved with the communication. Telephone networks have evolved on this model. Connection-oriented networks can work with switched channels (channel switching) or the message switching (packet switching or virtual connections).

Connection-oriented channel switching includes switching in the spatial domain (spatial separation of the channels – spatial switching) and in the time domain (time multiplexing of the channels).

Message switching consists of packet switching (a number of packets per message) and consignment switching (one packet per message).

A special position must be given to ATM switching, which is gaining in importance and will be described in Section 3.3.3.

Figure 3.2 – Overview of switching principles



3.1.3 References

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3.2 Channel switching

For channel switching, the relationship between the communication partners is implemented by connecting channels. After the relationship is created, the subscribers are directly connected with each other for the complete duration of the communication.

The spatial switched channel is the “classical” form of the connection. In the simplest cases, they are made with electrical connections, which are switched together with contacts. Switched channels can be either switched or fixed connections. For switched connections, the participating terminals are automatically connected together for a certain period time, based on the destination information of the source (using switching technology and signalling). Dedicated connections are created by network management measures for a certain period of time. The oldest network working on the connection-oriented principle is the telephone network.

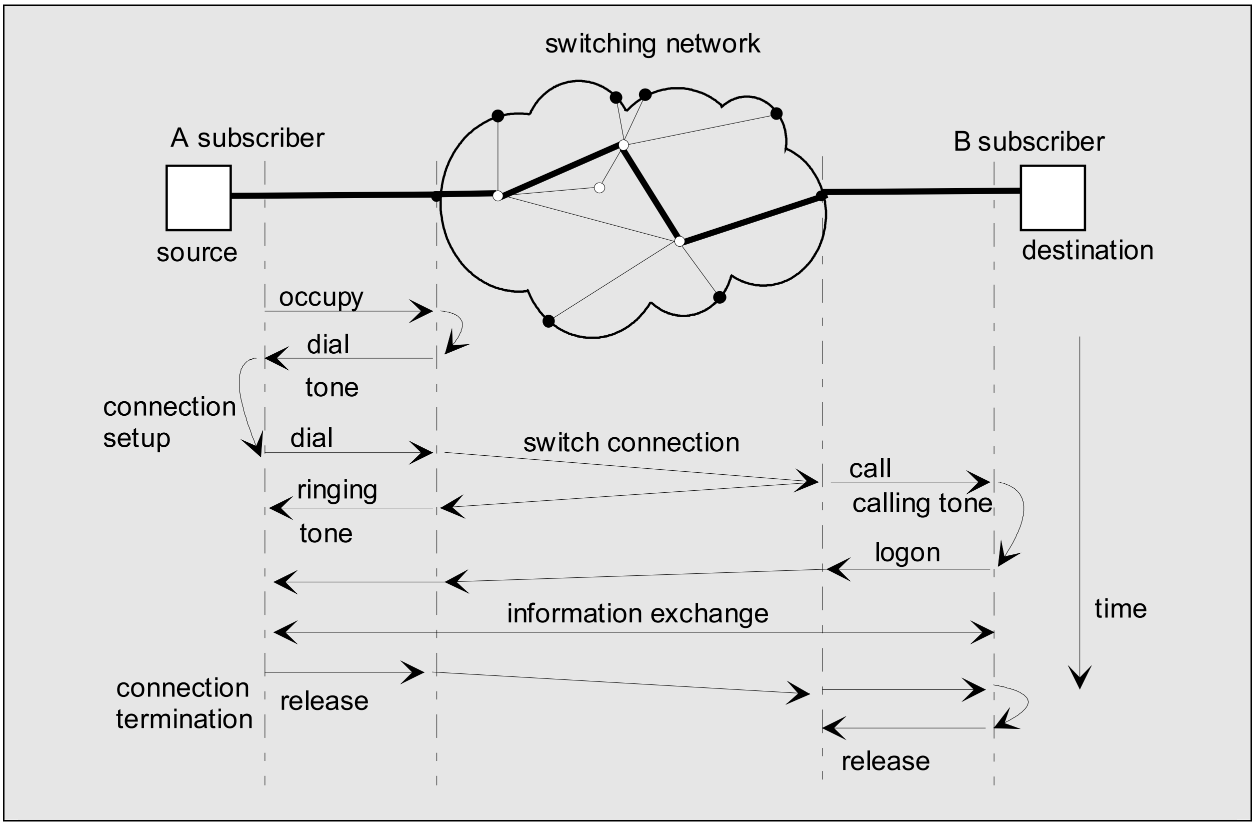
Spatial switching is the switching of physically separated electrical channels.

Time switching is the switching (rearrangement) of time slots in systems, in which the information from individual channels is transported in time slots.

Channel switching is also designated as circuit switching. For circuit switching, the creation of a connection is necessary before the actual communication is made; after the communication, the connection must be terminated again. Therefore the connection is divided into phases.

3.2.1 Connection phases

Figure 3.3 – Schematic representation of the phases of a circuit switchd connection



Connection set-up. The connection set-up is carried out by an exchange of signalling information between the active terminal equipment and the exchange, and between the exchanges. The initiative is taken by the terminal equipment which wants to set up the communication relationship (in telecommunication technology and in the above example in Figure 3.3: 'A'-subscriber). Thereafter follows the reservation of the switching device equipment to which the A-subscriber is connected. If this reservation is accepted, that is, if a facility is free to process the connection request, then the terminal equipment is informed (in the telephone network: using dial tone). Next, the terminal equipment notifies, by dialling, which other terminal it desires to connect to (dial information, address information). Then an attempt is made to establish a path to the destination terminal (B-subscriber). If this is successful, then the B-subscriber is called, and the A-subscriber is informed of the connection set-up (call display, in telephone network: ringing tone). After the B-subscriber has acknowledged the call (log on), the connection enters into the second phase. The created occupancy is, from the point of view of the A-subscriber, an outgoing call and, from the point of view of the B-subscriber, an incoming call.

In general, the requested connection extends over a number of switching configurations, and signalling is also necessary between them.

Information exchange. In the second phase of the connection the actual information exchange occurs which also can be accompanied by signalling. Thus, during the course of a connection, service components can be switched on and off and teleservices can be managed.

Connection release. The third phase of the connection is the connection release, which one of the terminals initiates by means of signalling. The switching equipment engaged and the occupied channels are released again. Data is collected for the recording of connection-dependent fees.

3.2.2 Structure of a switching system

Functional blocks. A switching configuration has a variety of functional blocks, which are either involved in or support the actual switching process:

- Switching: Connection of subscribers by means of subscriber lines and link lines, in order to create individual communication relationships.
- Administration: Administration of the subscriber lines associated with the exchange, trunk lines, the equipment of the exchange and the processes which run on this equipment. The collection and processing of fee and traffic data is also included.
- Maintenance: The ensuring of equipment availability of the central unit.
- Operation: communication between the central units and their operation personnel.

Figure 3.4 – Principle elements of a switching configuration from the point of view of the switching process

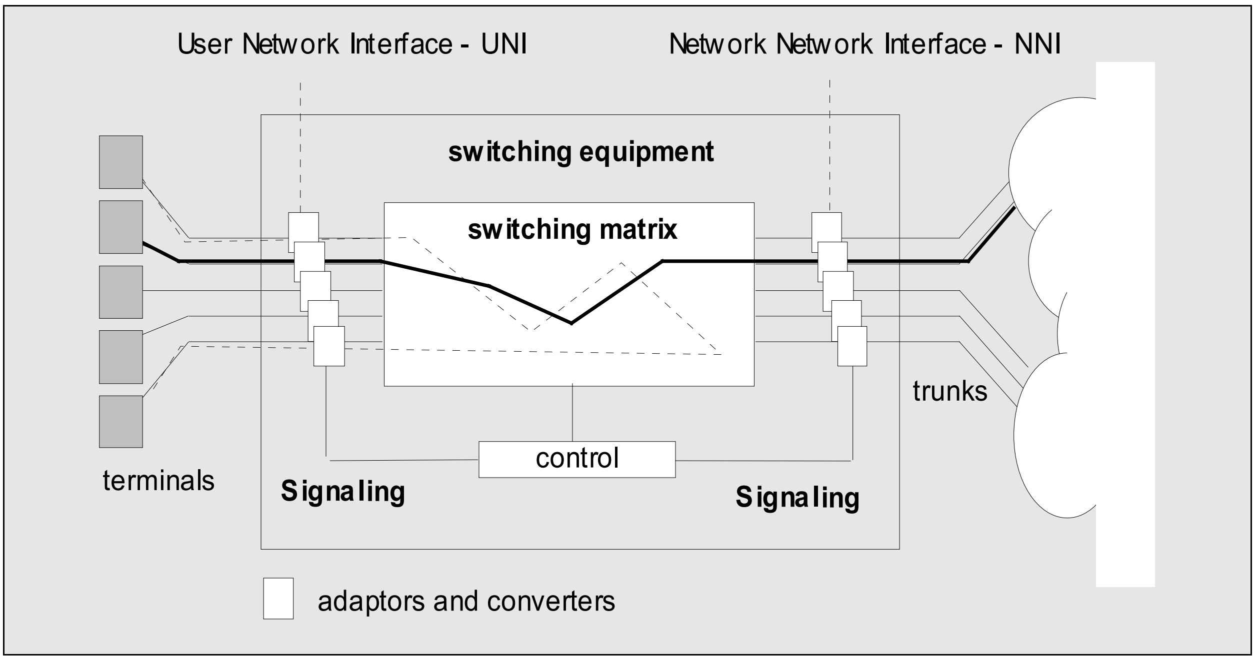


Figure 3.4 represents a local exchange. This is the most general case of a switching system, because here connections to subscribers, as well as connections to other exchanges, are represented. On the left side, subscriber lines connecting terminal equipment are represented, using the user network interface (User Network Interface – UNI). On the right side are trunk lines between the switching stations. Exchanges are connected by means of network interfaces (Network Network Interface – NNI).

A connection between two terminals attached to the same switching station is called an internal connection, and is represented with dotted lines in Figure 3.4. A connection from or to a subscriber, which is attached to another exchange is called an external connection. This kind of a connection is drawn in bold lines in the Figure.

Control An important element of the switching system is the control, which processes the signalling information from and to the terminal equipment and between the exchanges. The control system obtains the necessary information for adaptation from adapters and converters and from subscriber lines and trunk lines.

Switching matrix The actual creation of connections takes place in the switching matrix, also called switching network. It is the basic element of a switching system and is set up by the control system.

Periphery The periphery of the switching system must provide additional functionality so that the switching node can successfully integrate into the rest of the environment. The most important task requirements of this periphery are:

– the supply of power to the subscribers line, i.e. supplying the electrical energy,
– the protection of the switching system from electrical influences on the connections (for example, due to cable error, voltage overload, lightning etc.),
– the separation of payload and control signals for inband signalling (for example, from and to subscribers in a telephone network),
– the interference suppression of payload and control signals,
– the conversion of message forms (e.g. 2 wire, 4 wire conversion),
– recognition of incoming signalling,
– creation of signalling,
– recognition of errors for maintenance purposes.

The above functions are implemented in so-called trunk circuits and subscriber circuits. The subscriber circuit carries out the so-called BORSCHT function. BORSCHT is an English acronym for the functions:

– Battery (loading),
– Over voltage protection,
– Ringing,
– Signalling,
– Coding (e.g. analogue to digital (conversion),
– Hybrid (2-wire, 4-wire conversion),
– Test (error detection).

3.2.3 Task requirements of the function unit 'switching' of a switching system

For the task requirements of the most important functional units of a switching central unit, the elements of the service "switching" available to the user are described. The most important task requirements are:

- Search for a free unit for carrying out a function. Such a unit can be a free link in a certain direction (path seek), but also can be a software procedure instance for realising a service characteristic.
- Testing of identifications and access privileges.
- The occupation of a long-distance unit upon request. This unit is assigned to a connection to be created and locked for any other attempts at occupation.
- Switching on of dial tones.
- Receiving and evaluation of dialling information. Reception of dialling information and evaluation in terms of the selected direction, of the subscriber or of service characteristics.

- Signalling transmission, i.e. transmission of a telephone number from the switching system to another switching system or to terminal equipment.

- Connection, i.e. creation of a connection in the switching network.

- Connection termination, i.e. determination of fees, the signalling of the connection completion, release of the equipment.

- The disabling of a facility from use in case of malfunction, during maintenance or for other reasons (for example, to prevent traffic overload of other elements of the central unit or of the network).

- Release of allocated or disabled equipment within the exchange.

3.2.4 Switching matrix

The switching matrix is an arrangement of switching elements which are used to connect payload channels in a switching system.

The switching network is the central element of a switching facility. With switching networks, the required connections of transmission channels between the switching exchanges are created.

Based on the signalling information and available channels, the switching arrangement connects input ports and output ports. The task of the switching matrix is the set-up and release of connections, as well as handling the administration of the simultaneously existing connections.

In general, a switching network consists of a number of connecting stages. They are individual layers with a multiplicity of switching elements which are functionally parallel.

Function groups

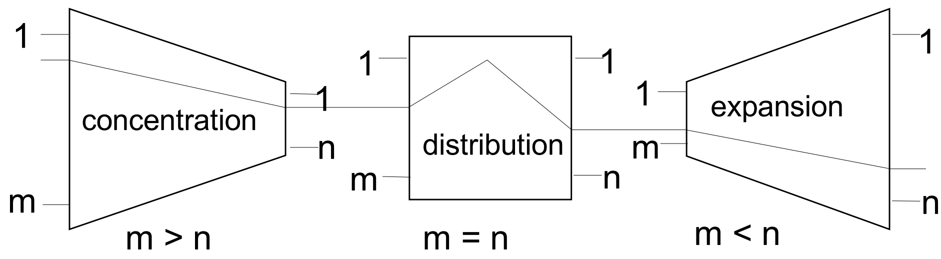
The complete switching network is divided into three important functional groups, in which the traffic to be switched is concentrated, distributed, and finally expanded. The most important function is the distribution of the traffic. The required technical equipment in general is very complex and can be better utilised with concentration. The concentration/distribution/expansion structure is functional. This basic structure of switching systems is the same for all principles that can be applied to switching, independent of whether it is switching between a variety of spatial connections, time slots or packets.

Concentration. Concentrating switching networks are used when more inputs than outputs are involved. Concentration is the switching of a number of input lines onto a few output lines. The traffic of the lightly utilised input lines is concentrated on more heavily utilised output lines. The expensive equipment assigned to the output lines is also better utilised.

Distribution. Linear switching networks are used when an equal number of inputs and outputs are involved. In distribution, the traffic is distributed according to its direction.

Expansion. Expanding switching networks are used when more outputs than inputs are involved. After distribution, the traffic must be reconstituted to the separate individual subscriber lines at the destination local exchange. The traffic is expanded.

Figure 3.5 – Concentrating, distributing and expanding in a switching network



A connection in a switching system is processed at first with a concentrating, then a distributing, and finally with an expanding, switching arrangement. This arrangement of the individual components of the coupling network is purely functional. For the practical realisation of switching network, a concentrating and expanding switching arrangement can comprise the same physical elements.

Spatially-separated switching matrixes

Spatially-separated switching is the oldest form of switching. A channel is made up of a certain number of lines (wires), which are connected with electrical contacts to one another. These contacts can be implemented by means of:

– relays,

– selectors (lift-rotate selector, motor selector),

– co-ordinate switches or

– electronic building blocks (transistors).

A switching matrix for three wires per channel and with 4 × 4 channels on the basis of a Strowger selector appears in Figure 3.6. An arrangement of three coupled mechanical switches represents one crosspoint.

Switching arrangement. The switching arrangement itself is a matrix, and connections can be created at the crosspoints. Figure 3.7 shows this kind of a coupling matrix in a so-called stretched representation. One crosspoint is required for a connection of an input to an output. Therefore, for m inputs and n outputs, m * n crosspoints are required. The switching network is free of blockage, which means that already existing connections cannot block new connections. Part a) of the diagram shows all coupling points, while the simplified representation in part b) of the diagram symbolises only the number of the inputs and outputs.

Figure 3.6 – Representation of the operating principles of a mechanical switching matrix

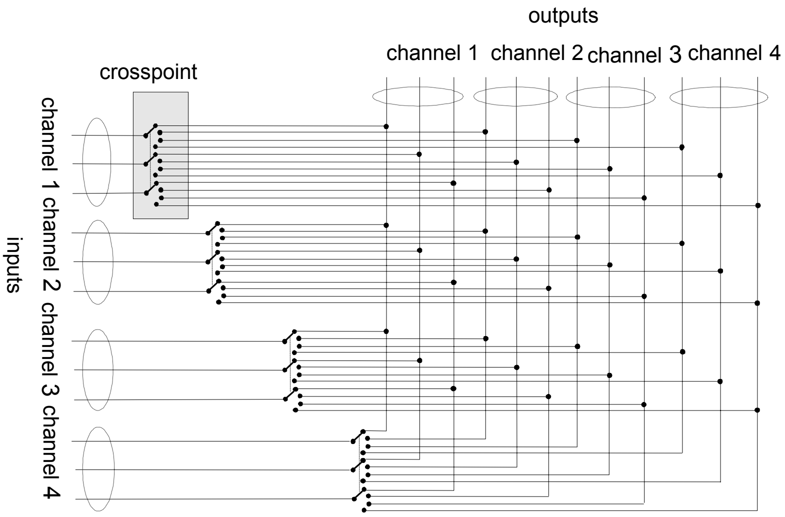
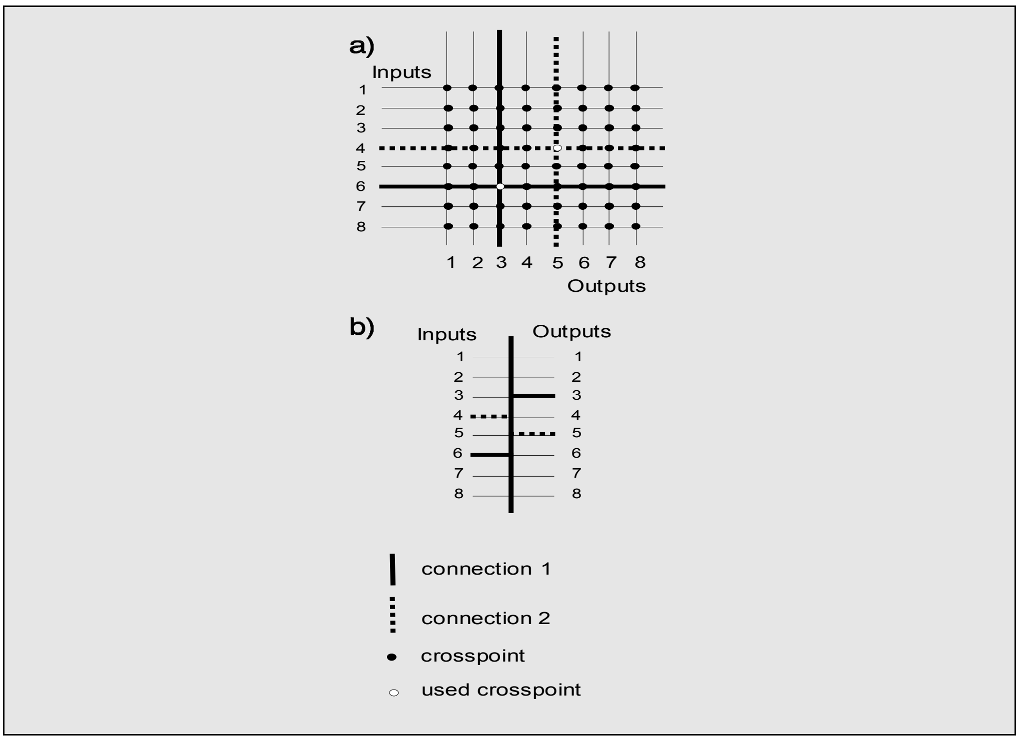


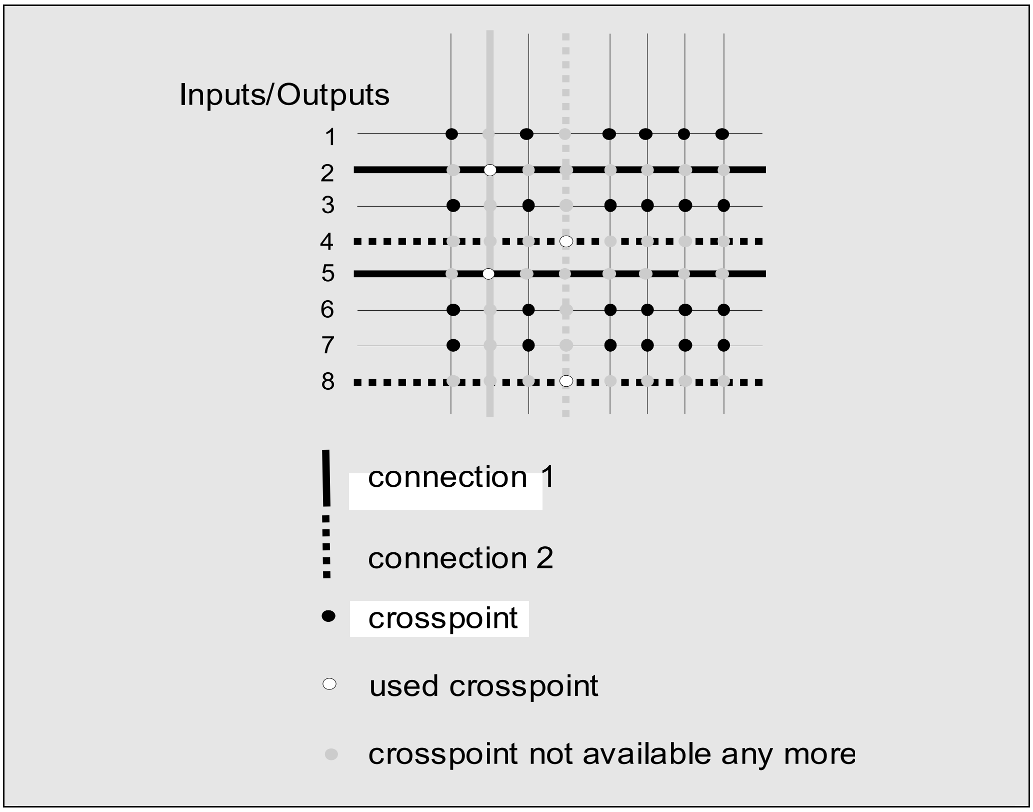
Figure 3.7 – Single-level switching matrix in a stretched arrangement
a) complete representation
b) simplified representation



Example: The coupling arrangement displayed in Figure 3.7 has m = 8 inputs and n = 8 outputs. Therefore m * n = 64 crosspoints are necessary. Every input can be connected with every output. Existing connections do not prevent other connections from being switched when other inputs and outputs are involved. In the example, connections exist between input 4 and output 5 as well as between input 6 and output 3.

Apart from the stretched arrangement, switching matrices can also be operated in the so-called reversal arrangement. In this case, inputs as well as outputs are connected on the same side (rows) of the matrix. The columns of the matrix serve to connect rows. For p columns of the matrix (m + n) * p coupling points are required. Two crosspoints are required for a connection. A maximum of p connections can exist at the same time. The disadvantage of this coupling matrix is that the connection between certain inputs and outputs cannot be created under certain conditions, because other connections already exist (internal blockage).

Figure 3.8 – Switching matrix in reverse arrangement



Example: The coupling arrangement shown in Figure 3.8 has m + n = 8 connections which could be inputs or outputs. Determined by p = 8 columns of the coupling matrix, p * (m + n) = 64 crosspoints are necessary. Every connection uses a column of the matrix (in this case, drawn in grey) to complete the circuit. Therefore a maximum of p connections can be switched. Every switched connection effects that the coupling points of the rows and columns required for the completion of the circuit cannot be used for other connections. The coupling points no longer in use are also drawn in grey.

This configuration of the coupling matrix meets an important requirement for the configuration of switching matrixes: the number of the employed technical elements should be approximately proportional to connection capacity; this not the case for a coupling matrix in a stretched arrangement, in this case it is a quadratic dependency.

Because of the necessary requirement for extensibility, switching networks should be modularly designed. This can be achieved by dividing up large switching matrixes into smaller matrixes and then switching these matrixes together over a number of levels. With multi-level switching networks and the switching together of smaller matrixes, fewer crosspoints are required than for single-level switching networks. But in the case of multi-level switching arrangements, internal blockages are possible. The probability of an internal blockage goes up with the concentration factor of the switching matrix and declines with the size of the individual switching matrix.

Example: The switching network, which is represented in Figure 3.9, allows for the connection of up to 100 subscribers. A maximum of five internal connections can be simultaneously set up, as well as up to three external trunk groups with up to five connections each.

For the case m = 10 inputs and n = 5 outputs, per switching matrix in a stretched arrangement in layer 1 m * n * 10 = 500 crosspoints are required.

In layer 2, the switching matrices also have m = 10 inputs and n = 5 outputs. The 5 switching matrices of this level thus have a total of 10 * 5 * 5 = 250 crosspoints.

In layer 3, the number of the crosspoints can be calculated from m = 5, n = 5 and the number of matrices which is five. This yields 5 * 5 * 5 = 125 crosspoints.

In total, 875 crosspoints will be required.

Hence because of internal blockage, it is not possible to create more than five connections for a subscriber group out of 10 subscribers which belong to one and the same switching matrix of the first layer. Furthermore, not more than five internal connections, and not more than five connections to the external trunk group, can be created simultaneously.

Two connections are displayed:

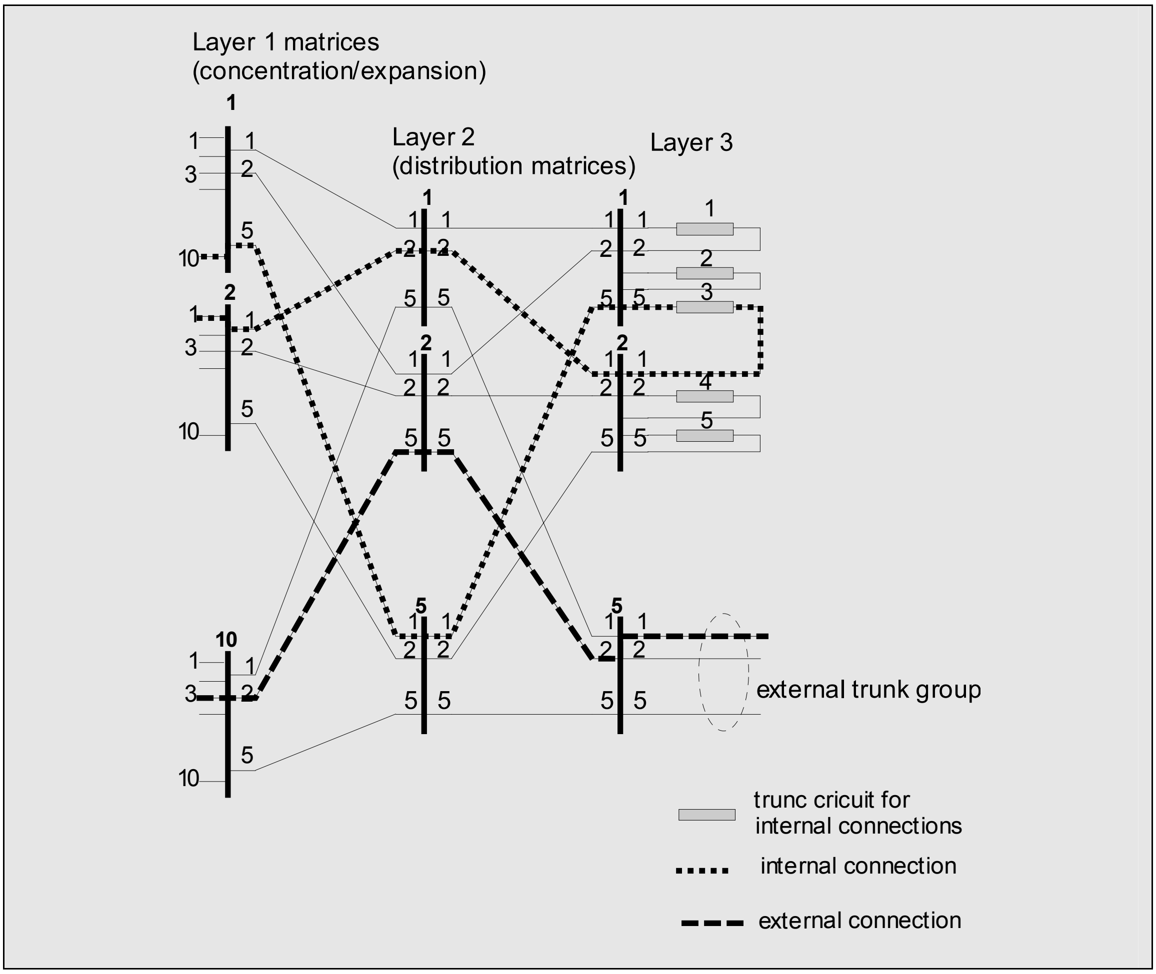
– line 10 of the first matrix of layer 1 connects to line 1 of the second matrix of layer 1 (internal connection) and;

– connection 3 of the 10th matrix of layer 1 connects to line 1 of the external trunk group (external connection).

By carefully designing the switching network layers and the connections between the layers, a compromise can be found between crosspoint number and blockage probability. This information on switching networks mainly refers to the switching of spatially separated channels, which can be implemented with Strowger selectors or co-ordinate switches.

But channels can also be in different forms. It is possible to assign a channel a fixed carrier frequency and switch this carrier in the switching system. Another possibility is the assignment of a time slot to a channel. In digital switching technology, the spatial and the temporal domains are utilised. In switching devices, spatial and time switching arrangements are often used in combination.

Figure 3.9 – Multi-layer switching network

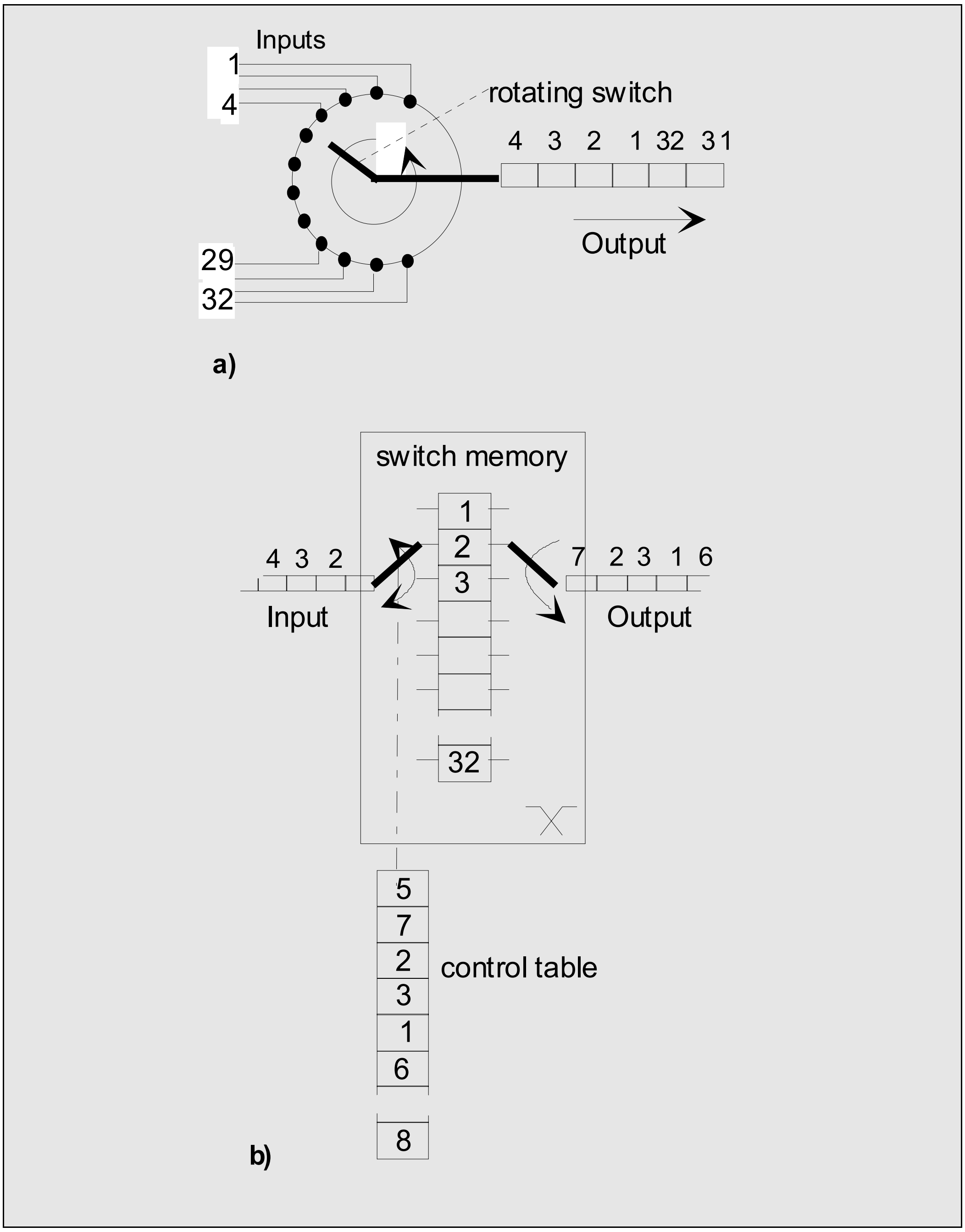


Time-division switching networks

Time-division synchronous channel splitting. In the case of time-division channel splitting, individual time slots are assigned the information to be transmitted in the channels. This technology, for example, is applied for Pulse-Code-Modulation (PCM). The assignment of individual channels to time slots is shown in Figure 3.10. The assignment is rigidly defined in a frame structure. The position of individual bits in the frame determine to which information relationship they belong. The synchronization which is carried out for a frame must last for the time it takes for a complete pass through the frames. A time frame is represented in Figure 3.10 a) as a complete cycle of the rotating switch. 32 channels are nested in it and a cycle requires 125 ms.

Time-switching arrangement. The principle of a simple switching arrangement for switching the time position of an individual channel is shown in Figure 3.10 b). In this case, the information which arrives at the input in individual time slots is written to specific fields of the switching memory by a controller and temporarily stored. This writing process is controlled by a control table. The reading of information from the memory occurs in a fixed sequence. The control table contains the assignment of the time slots of the output lines to those of the input lines. It is also conceivable that the data is written in a fixed sequence and read out with a control table.

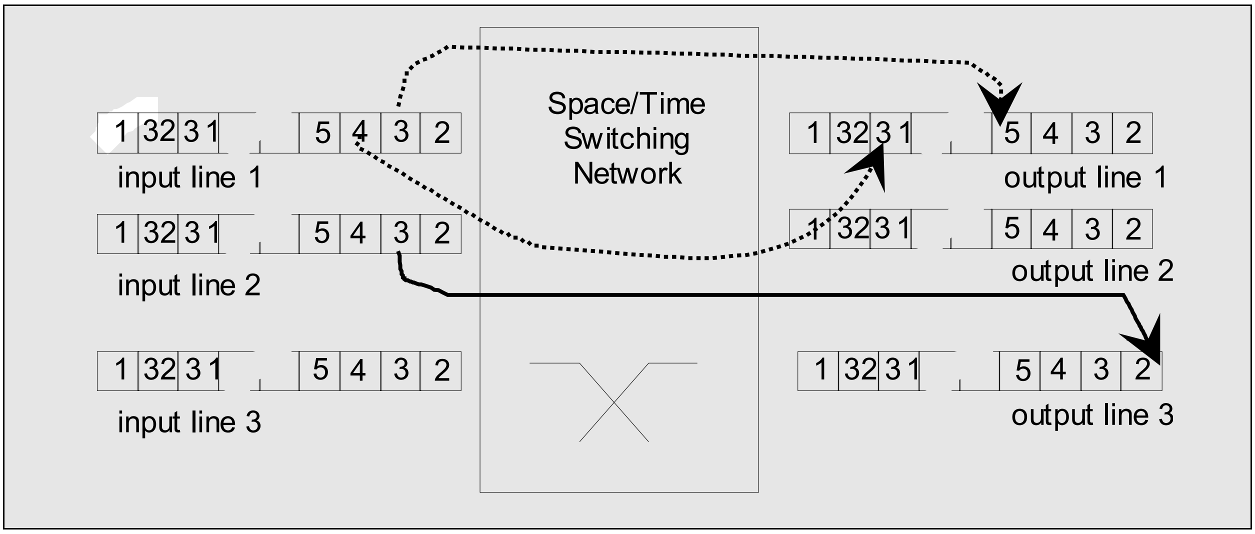
Figure 3.10 – a) Assignment from channels to time slots
b) Rearrangement of time slots because of intermediate storage



In every case, storage of the time slot information is required for the rearrangement of time slots. This can occur at the input of the coupling field, at the output of the coupling field, centrally for the complete coupling matrix, or distributed for every coupling position. For a detailed representation of the storage types, refer to the section on ATM switching, because the same principles will be applied there.

Time/Space Switching. In general, a number of PCM input lines reach a switching network. The job of the switching network consists of executing the rearrangement of the time slots as well as co-ordinating between PCM connections. For this, a space and time switching network (Figure 3.11) is required.

Figure 3.11 – Spatial temporal switching



Example for Figure 3.11: Time slot 3 of the input line 2 is to be assigned to time slot 2 of the output line 3 (solid line). For this task, a temporal and a spatial switching process are necessary. The dotted-line rearrangements, in contrast, require only time switching. Technically, the spatial and the temporal switching can be carried out at the same time. For this purpose, all spatially-separated input lines on a line are multiplexed (note: for inputs, this line must have more than an n-fold processing speed) and stored; the individual spatially-separated output lines of the coupling arrangement, parallel to each other, are read out of the correct time slots from the common memory.

3.2.5 Control of switching devices

The special feature of the switching device control system is that a connection almost always pass through a number of network nodes and therefore a number of switching stations, and all of these switching stations are incorporated into the control system of the connection.

The transmission of control information between switching stations and from/to the terminal equipment is carried out by signalling.

Every connection is built up piece by piece by selecting channels. This selection subdivides into:

– a forced selection, which determines the direction in which the connection will continue to be built, and

– a free selection, which automatically dials up a free channel in this direction,

– the forced selection is always controlled by the dial information.

The dial information required for the control system of the participating switching device is created in the calling terminal. If this dial information is used directly to control the switching system, this is called direct control. If the dial information first goes to temporary storage and then is evaluated, this is called indirect control.

The direct control system was introduced with the introduction of the lift-rotate Strowger selector. The impulses of a dialler directly control the lift steps. In the pause between two dialled digits, the free selection of a channel in the selected direction can be carried out. The next dialled digit now directly controls a selector in the next selection level or in another switching station.

The indirect control system has applications mainly in SPC switching and computer-controlled switching systems.

The direct control system is no longer used today. The indirect control has the following advantages:

- Before individual segments of a connection become occupied, it can be determined if a path can be found through the network up to the destination terminal equipment, thus avoiding the stepwise occupancy of channels before the actual effective connections can be completely made;
- Considerably more complex methods of path searching (routing) for a connection through the network can be applied than with the stepwise connection set-up.

3.2.6 References

ITU-T Reference exchanges – Introduction and field of application

[Q.511] (11/88) – Exchange interfaces towards other exchanges

[Q.512] (02/95) – Digital exchange interfaces for subscriber access

[Q.513] (03/93) – Digital exchange interfaces for operations, administration and maintenance

[Q.521] (03/93) – Digital exchange functions

[Q.522] (11/88) – Digital exchange connections, signalling and ancillary functions

[Q.541] (03/93) – Digital exchange design objectives – General

[Q.542] (03/93) – Digital exchange design objectives – Operations and maintenance

[Q.543] (03/93) – Digital exchange performance design objectives

[Q.544] (11/88) – Digital exchange measurements

[Q.551] (11/96) – Transmission characteristics of digital exchanges

[Q.552] (11/96) – Transmission characteristics at 2-wire analogue interfaces of digital exchanges

[Q.553] (11/96) – Transmission characteristics at 4-wire analogue interfaces of digital exchanges

[Q.554] (11/96) – Transmission characteristics at digital interfaces of digital exchanges

[Q.700] (03/93) – Introduction to CCITT Signalling System No. 7 (Series, Q.700 – Q.788)

[Q.920] (03/93) – Digital Subscriber Signalling System No. 1 (DSS1) – ISDN user-network interface data link layer – General aspects (Series Q.920 – Q.957)

[Q.1200] (09/97) – General series Intelligent Network Recommendation structure

[Q.2010] (02/95) – Broadband integrated services digital network overview – Signalling capability set 1, release 1

3.3 Message switching

In the case of message switching, no channels are established on which the information is exchanged, but rather individual messages units, most often packets, which contain all or a part of the information to be transmitted, are switched.

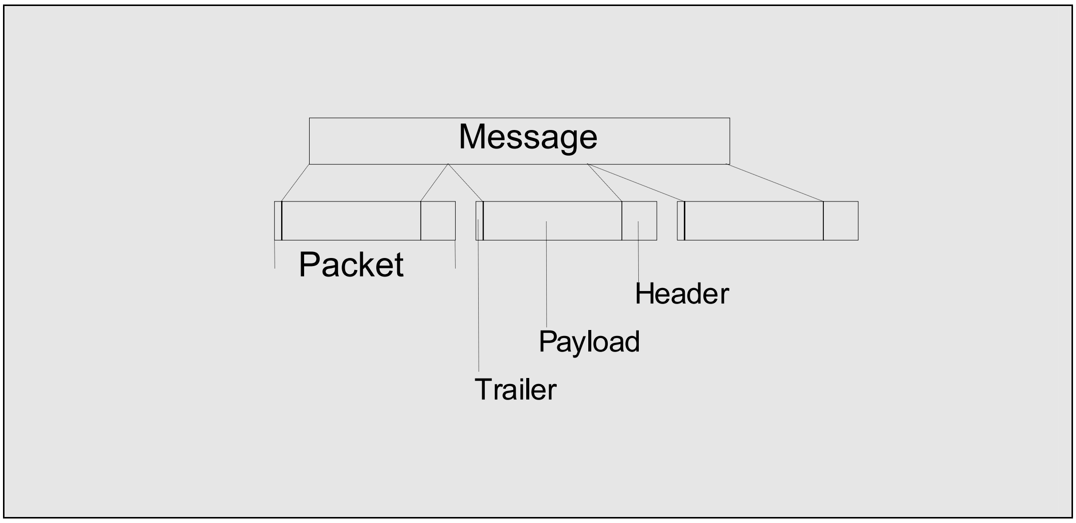
This occurs exactly like one would imagine the "switching" of postal letters in a network of post offices: The packets are supplied with addresses which give information about the receiver. In each switching station, the address is evaluated and the message is forwarded in a direction which brings it closer to its destination. The switching is carried out separately for each individual message unit. Therefore no connection set-up is required. Packets, which belong to the same information relationship, can take different paths through the network.

Store and forward switching. Message switching is often called store and forward switching. Typical for this configuration is that the packets are lead step for step (from switching system to switching system) through the network. The packets are stored temporarily in each of the network nodes.

3.3.1 Packet switching

Packet switching switches information that is divided into a number of packets. A packet in this sense has the following basic set-up:

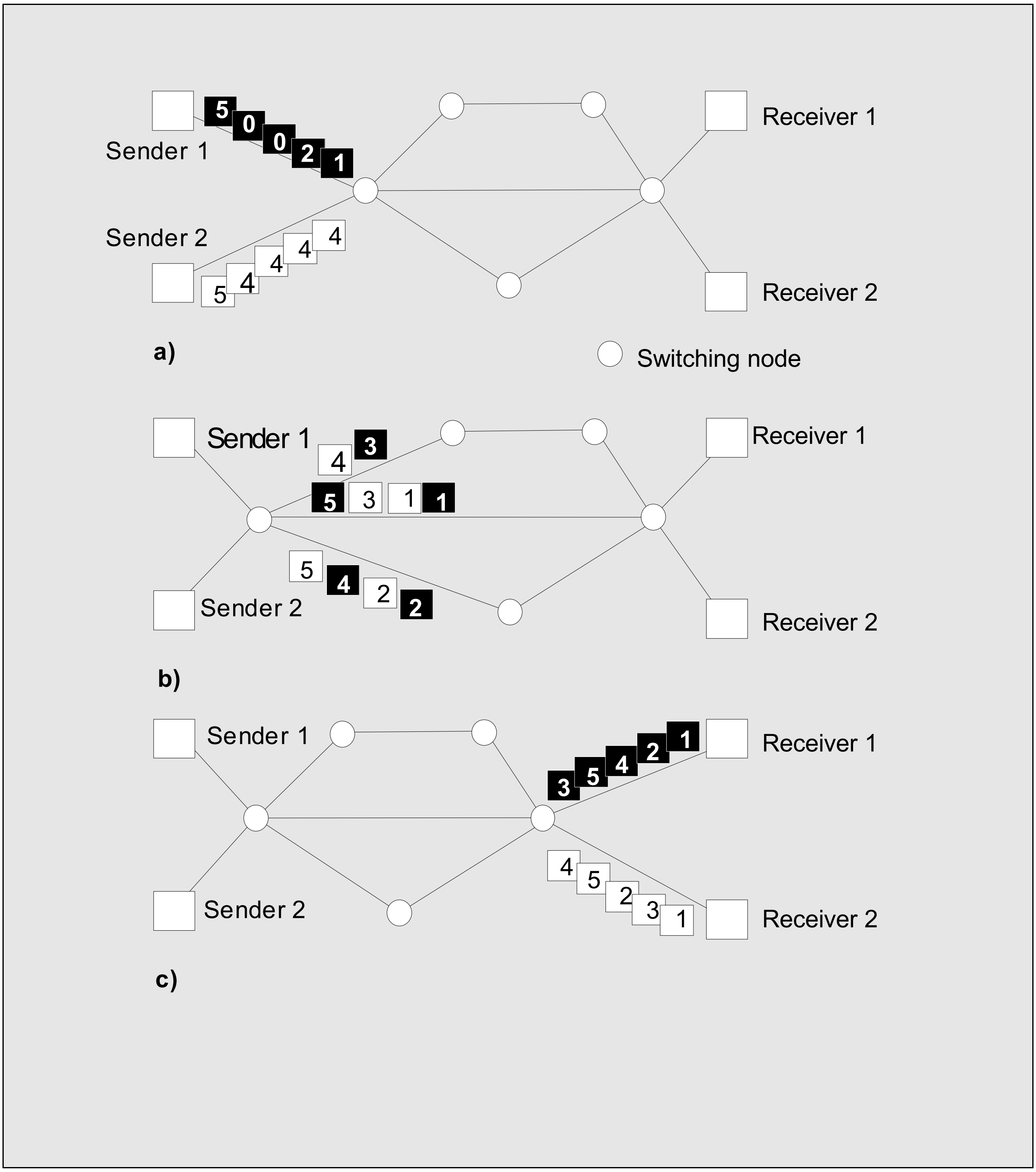
Figure 3.12 – Set-up of packets in packet switching



Packet. A message is divided into a number of units. These units are supplied with a header and a trailer. The header, payload and trailer form a packet. Packets can be of fixed or variable length. The packet trailer is not necessary for certain switching procedures.

The packets are created at the transmitting terminal equipment. At the network nodes, the addresses of the packets are analysed and are forwarded in a direction which will bring them closer to their destination. For this purpose, packets need not necessarily take the same path. The forwarding process is dependent on the traffic load which is currently on the network.

Figure 3.13 – Switching of packets in a packet switching network
a) phase 1: transmission of the packets
b) phase 2: switching of packets to a network node
c) reception of the packets by the receiver



Comment to Figure 3.13: The simultaneous but independent transport of two message units is described. At first, both transmitters allocate the transmission information into the packets 1 to 5. These are passed on to the network in the order of their numbering (Figure 3.13 a)). The first switching node test attempts to direct the packets on the shortest path in the direction of the receiver. Both receivers are connected to the same switching node. The expedition of the packets is first of all successful for both of the first packets. Now the transmission capacity on the direct connection to the receiver is exhausted for the moment and so the respective second packets are sent over the alternative lower part of the network. With this transmission, this path is also fully utilised. The third packet of the information relationship 1 must now be sent on a longer alternative along the upper part of the network, because now the first alternative also has no further transmission capacity available. Now a packet along the direct path can be accepted (packet 3 of information relationship 2), the next packet (packet 4 of information relationship 1) is once again sent on the shortest alternative. Packet 4 of connection 2 takes the long alternative. Once more a packet can be sent along the direct path and the last packet (packet 5 of the relationship 2) can take the short alternative (Figure 3.13 b). Because of the different transmission times for each route, the packets arrive at their receivers in the order shown (Figure 3.13 c).

The advantages of packet switching are:

– rapid transmission without connection set-up times, especially appropriate for short, sporadic information transmission and a low number of packets,

– good time and space capacity utilisation of the network resources, especially for sporadic, burst-mode traffic.

The disadvantages of packet switching are:

– transmission time varies and cannot be guaranteed,

– resource cannot be guaranteed (bandwidth),

– packets can overtake each other (see Figure 3.13 c)),

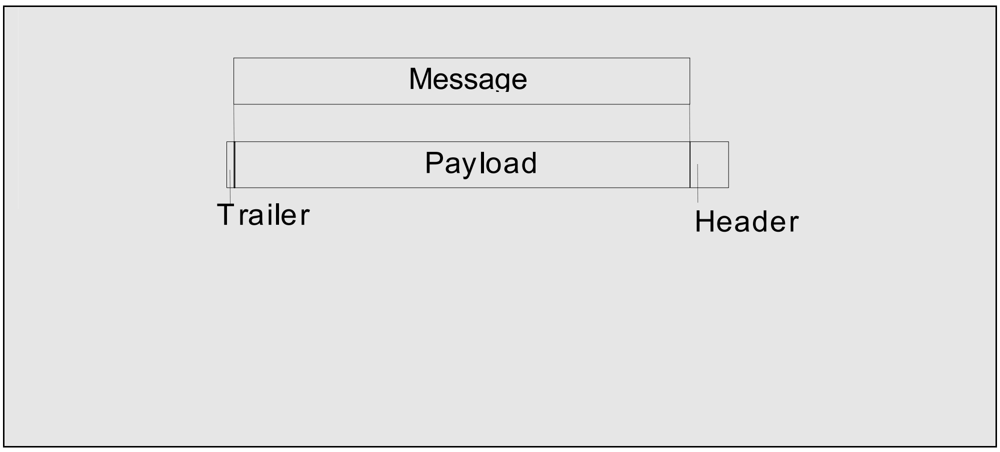
– higher computing power requirements for the routing of the packets.

3.3.2 Message switching

Message switching conveys packets which contain the complete contents of an information relationship.

A message packet which is conveyed in transmission switching, has the following design.

Figure 3.14 – Set-up of a packet for message switching



The packets have a variable length. The complete contents of a message are contained in a packet. Therefore, in contrast to packet switching, there is no need for the division of the message into data blocks and the protocol overhead that results. The process does not differ from packet switching from a technical point of view. It is used, for example, for the short message service (SMS) in GSM networks.

3.3.3 ATM switching

In the case of ATM switching, the composition of information packets is similar to that for packet switching. They all have the same length of 53 bytes. All packets of an ATM connection take the same path through the network, for which the transmission capacity has been reserved in advance.

ATM switching differs from classical packet switching by the constant packet length and the determination of a connection path. This allows the switching of ATM cells to be simpler and computationally easier to control.

Storage principles

A requirement for the switching of ATM cells is that the cells in every switching system are temporarily stored. For this purpose, the following basic principles can be applied:

- Input memory: Per input, the incoming cells are stored in memory on the principle first-in-first-out (FIFO). For the switching process, an internal blocking-free matrix is employed. The disadvantage of this storage method is the possible blockage of waiting cells in the FIFO, so that even though the respective output is free, it is possible that a cell must wait for switching because previous cells to other outputs must be handled first.

- Output memory: Immediately after arriving, the cells are switched to a FIFO per output, and read out from there with the output line cycles. On the input, only the storage of one cell per lead is necessary. The disadvantage of this storage method is that the internal speed of the switching matrix must be greater than the speed of all incoming cells.

- Central memory: All incoming cells are stored in a common memory. This can be smaller than the sum of all separate memory requirements, but the control system for memory access is complex and very high-speed memory access is required.

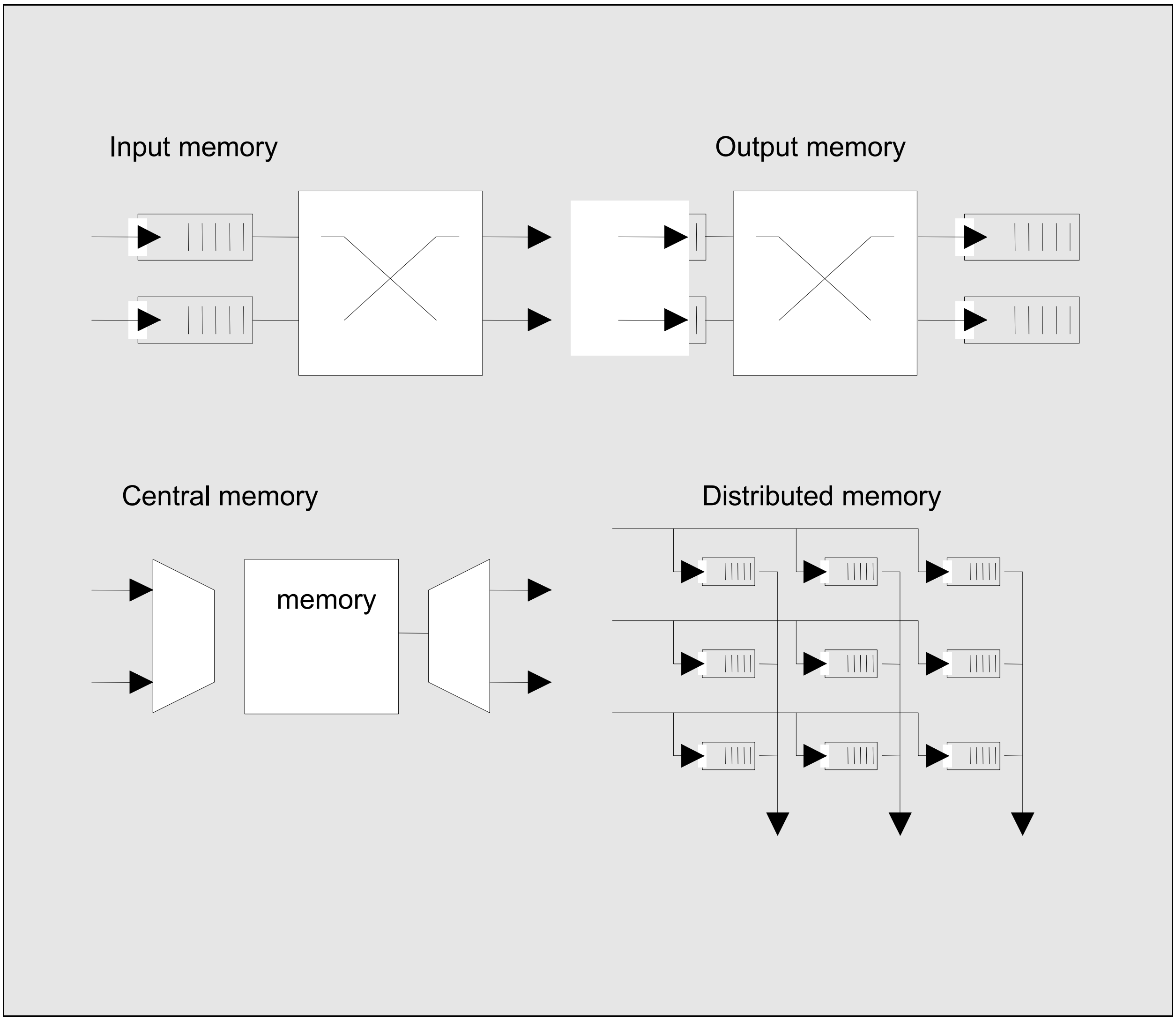
Distributed memory: In a matrix made up of input and output lines, memory is allocated at every crosspoint to allow the multiplexing of the cells on the output lines. The disadvantage of this method is the large memory requirement.

3.3.4 Virtual connections

In the case of virtual connections, individual packets are switched, but all packets of an information relationship are transmitted along only one path which is established at connection set-up.

Connection orientation. Before the information exchange begins, there is a connection set-up which determines if a path with adequate transmission capacity is available between source and sink. This channel is not occupied during the total connection time, but only when the transmission capacity is required. If no packets are available for some duration, the transmission channel can be used for other virtual connections. The capacity of transmission sections can even, within certain limits, be overbooked (statistical multiplex gain), nevertheless, all virtual connections have access to guaranteed resources and at times even have the use of more bandwidth than they were guaranteed.

Figure 3.15 – Storage principles in ATM-switching

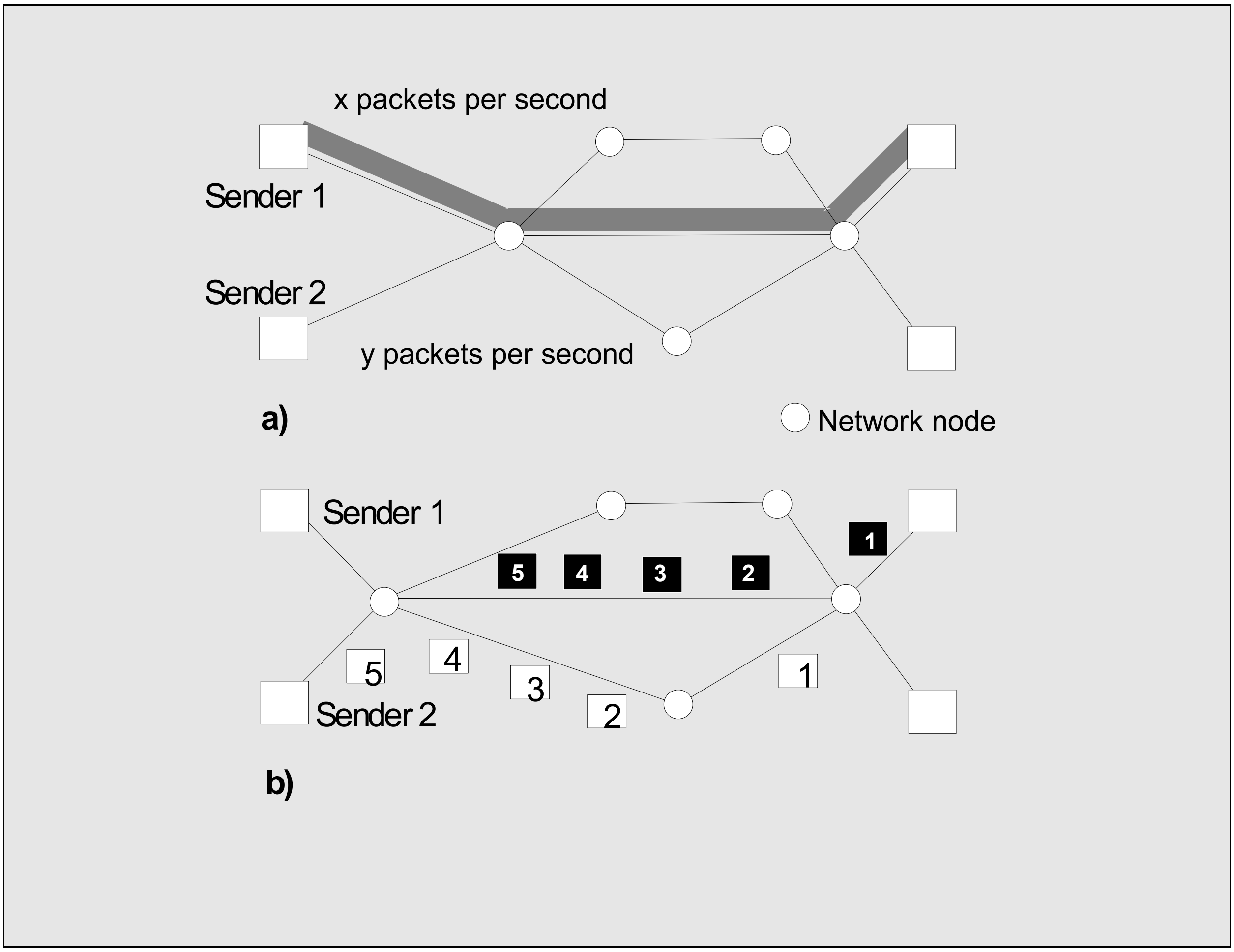


Virtual connections combine the advantages of packet switching and channel switching. They:

- do a good job of utilising the resource of the network (an advantage of packet switching);
- can quickly make available large transmission capacities (an advantage of packet switching);
- guarantee resources (an advantage of channel switching), and;
- have a control system which is inherently less complex to realize than with a strict packet switching system.

In phase 1 (connection set-up), the transmission capacity along both designated paths is reserved. In order to guarantee the desired bandwidth, as in the example, both connections must be led along different paths. For the transport of the packets in phase 2 (information exchange), the reservation of path and bandwidth of service quality (Quality of Service – QoS) ensures that packets cannot overtake each other and are delivered within the timing requirements.

Figure 3.16 – The switching of packets in a switching network with virtual connections
a) phase 1: connections set-up
b) phase 2: switching of the packets along the set paths



3.3.5 Switching and routing

Switching

Switching is the creation of connections in a classical telecommunications network for a limited period of time by the interconnection of channels (line or circuit switching). During connection set-up, which is carried out before the actual information transmission occurs, the creation of the connection is controlled by signalling. Connections can also be virtual as is the case with ATM.

Switching is carried out at layer 2 of the OSI Reference Model.

Routing

Routing is the directing of data packets, based on the complete address of the destination of the sender contained in the data header, to the receiver over a varying number of nodes (routers) through the network. The job of the routing function is, for example, to transport datagrams in a packet network from a transmitter to one (unicast) or numerous (multicast, broadcast) destinations. For this, two sub-tasks must be performed:

– the construction of routing tables, and;

– the forwarding of the datagrams using the routing tables.

The routing process described here is the forwarding of data packets. It has nothing to do with path searching for switched circuits under certain network conditions, such as in the case of overload, errors, or for optimising the costs of a connection (least-cost routing).

The datagrams are transferred from one router (next-hop) to the next (hop-by-hop). A given router knows the next router which lies in the direction of the destination. The decision on the next router (next-hop) depends on the destination address of the datagram (destination based routing). An entry in the routing tables contains the destination and the next-hops that belong with it, as well as supplementary data.

The routing table determines the next node that a data packet must reach in order to get to the desired destination. Routing tables can be:

– static, or

– dynamic.

In the case of static routing, the next-hop of a route is entered as a fixed location in the tables. Static routing is appropriate for smaller networks and networks with a simple topology. In the case of dynamic routing, the next hop is determined from network state information. Employment makes sense for larger networks with a complex topology and for the automatic path adaptation in case of error (backup), and in case of the overloading of the network parts.

3.3.6 References

ITU-T References

[I.232.1] (11/88) – Packet-mode bearer service categories: Virtual call and permanent virtual circuit bearer service category

[I.232.2] (11/88) – Packet-mode bearer service categories: Connectionless bearer service category

[I.232.3] (03/93) – Packet-mode bearer service categories: User signalling bearer service category (USBS)

[I.233] (10/91) – Frame mode bearer services, ISDN frame relaying bearer service and ISDN frame switching bearer service

[I.233.1 Annex] (07/96) – Frame mode bearer services: ISDN frame relaying bearer service – Annex F: Frame relay multicast

General References

Schwartz, M.: Telecommunication Networks. – Reading: Addison-Wesley, 1988

3.4 Telephone switching technology

Telephone switching technology is the technical basis of what is applied for the switching of connections in analogue and digital networks for the telephony service and in ISDN. It is characterized by the switching of narrow band channels.

The telephone network is the oldest telecommunication network in the world. The first switching functions were also introduced into this network.

Table 3.1A – Development of the telephone switching technology

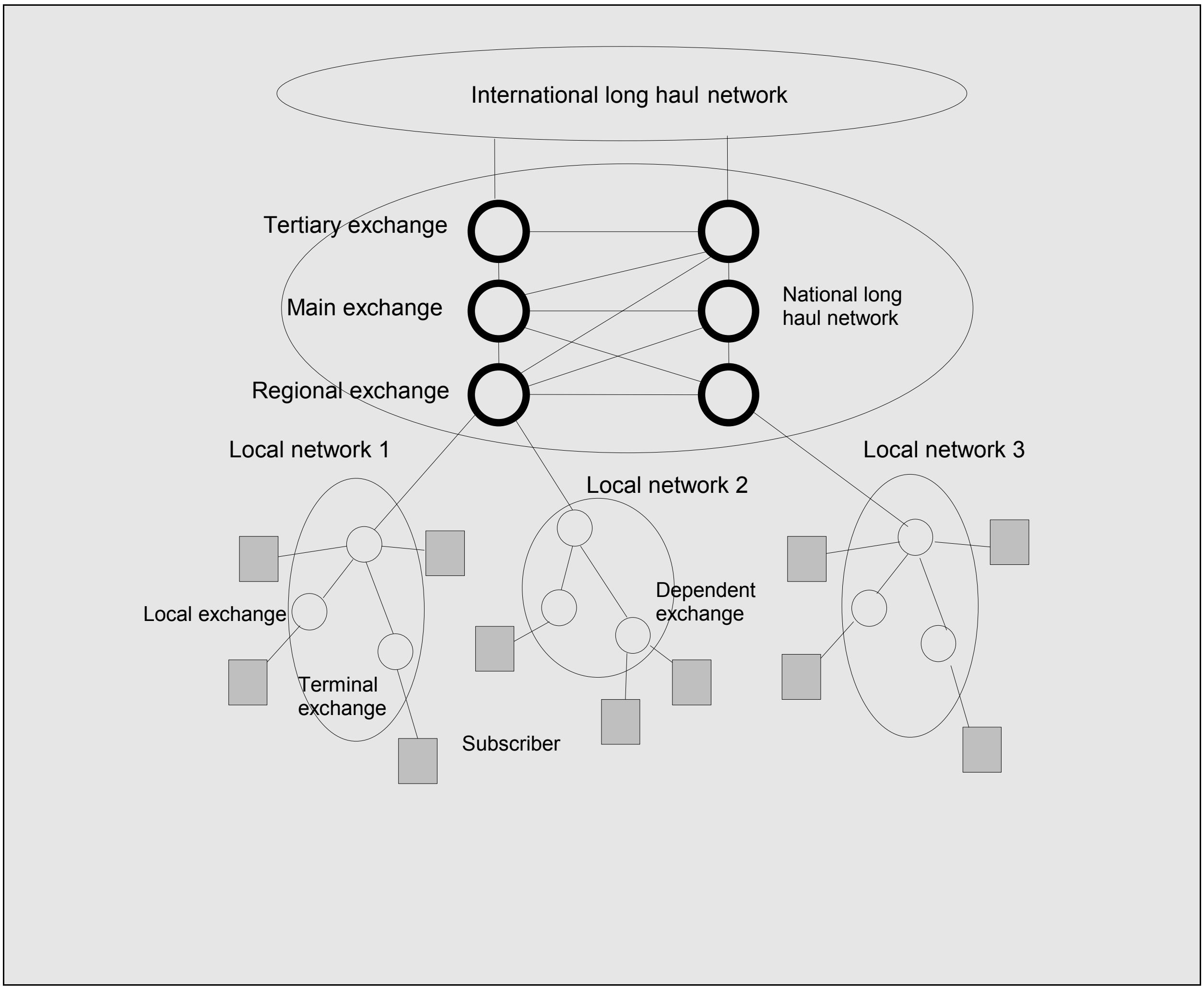
1877	First telephone switching (manual switching in USA)
1892	First automatic switching (USA)
1965	First fully electronic local switching system (USA)

Table 3.1B – Development of the telephone switching technology in Germany

1881	First telephone exchange in Germany (Berlin, 8 subscribers)
1908	First automatic switching in Europe (Hildesheim, 900 subscribers)
1923	First fully automatic switching beyond the local region (Weilheim)
1970	Total-area coverage self-dialling service in Germany
1975	Computer-controlled local switching technology in Germany
1984	First digital remote switching station in Germany
1985	First digital local switching station in Germany
1998	Completion of the total digitalization of the telephone network in Germany

The worldwide telephone network today has a structure as shown in Figure 3.17.

Figure 3.17 – Structure of the worldwide telephone network



3.4.1 Local network

In the lowest level of the telephone network is the local network to which the subscriber is connected. It is made up of local exchanges, terminal exchanges and dependent exchanges which are remotely controlled from local exchanges.

Local networks can be of different sizes. While on the one hand, digital concentrators can be employed for very small local networks with up to a few hundred subscribers, if the number of subscribers is a few thousand then remote-controlled switching stations are employed. Very large local networks can have up to 100,000 subscribers. They are implemented with independent local exchanges.

The subscriber is connected to the local network by means of subscriber lines. The local exchanges are tied together by local trunk lines.

3.4.2 Long haul network

Local networks are connected through national long haul networks. These are mapped by the regional exchanges, main exchanges and tertiary exchanges.

This structure can also be expressed in the subscriber numbering; i.e. within a local network, only the telephone number of the subscriber is selected in order to connect to a another subscriber in the same local network. From outside the local network, the user must dial the local network code and furthermore, for a subscriber in another country, the country code.

Local networks and long haul networks internally can contain a number of hierarchical levels; in some countries, though, no difference is made between the local and the long-distance level.

It is possible, that the actual path that a connection takes in the network does not follow the hierarchy set by the numbering. By means of so-called traffic routing, shorter and therefore more efficient paths are possible. Digital, computer-controlled telecommunications systems contain numbering schemes that are independent of the hierarchical structure of the network.

The national long haul networks of the individual countries are again networked through the international long haul network. This is subdivided once more into two network levels: the intercontinental long-distance network has exchanges in New York, London, Sydney, Moscow and Tokyo. The sub-level is constructed by the continental long-distance networks. The continental long-distance networks have the following codes:

1) North America;

2) Africa;

3) and 4): Europe;

5) Central and South America;

6) Australia, Oceania;

7) Russian Federation;

8) Asia without Russia, India and the Arabic countries;

9) India and the Arabic countries.

3.5 Connectionless messages transfer

3.5.1 Principles

In connectionless message transfer, the transfer is carried out in packets that include both the source and the destination addresses. All packets reach all network nodes and terminals of the respective network. Every receiver looks for and retrieves "his own" messages based on the address information given.

This form of message transfer is especially used in networks for data transmission, for example, LAN or WAN applications. The advantage of this method lies in the ability to send information without previously setting up a connection. Additionally, no routing mechanism is required. This is especially advantageous for sporadically occurring, short information relationships.

The transmission is possible only in frames or packets. Since the packets contain source and sink addresses, no connection set-up and termination is required. The packets are transmitted spontaneously. But the availability of sufficient resources in the entire network cannot be guaranteed, nor whether the sink has the ability to accept the transmitted information. Therefore measures are required to ensure that a message has really reached the sink.

This is implemented with protocols, at higher levels of the OSI reference model.

A shared medium is a transmission medium that is used by a number of communication relationships. The transmission capacity for a specific connection is dependant upon the traffic of all other communication relationships.

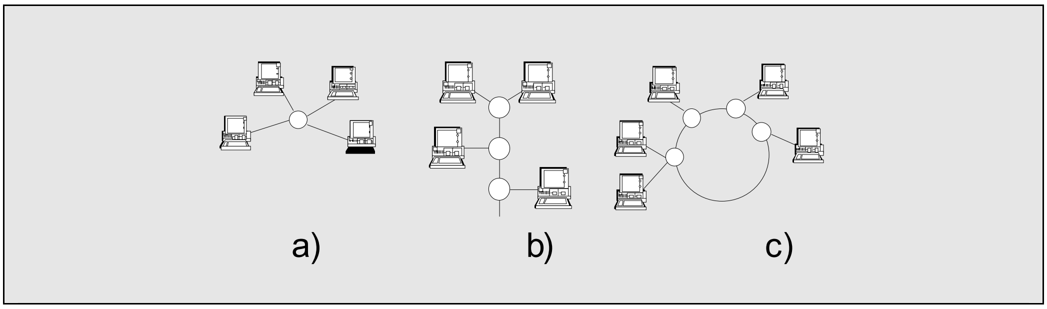
Media access. Since no connections for individual information relationships have been made, all existing information relationships must share the transmission medium (shared medium). For this reason, there is always a time frame and regulation for the media access. This can either be based on chance and uncoordinated, i.e. access is not previously agreed upon with other stations (random access), or the stations are given transmission rights at predetermined time slots (token access).

Network topologies. Figure 3.18 shows the possible network configurations for connectionless message transfer. One can see that no hierarchical composition of the network is possible as would be the case with tree or meshed networks.

The interconnection of connectionless networks, which would imply the creation of hierarchies, makes it necessary to selectively make a distinction between internal traffic (source and sink are contained in the same network) and external traffic (source and sink are located in different networks). For this purpose, bridges and routers have become typical network elements of LANs and WANs. They analyse the address information of the data packets and filter the external traffic for the transfer to the next higher network level.

If connectionless service in networks with connection set-up is offered, special network nodes (servers) are required which accept connectionless traffic and after analysing the address information, pass it on. This causes a logical sub-network of fixed address connections to be created for the connectionless traffic.

Figure 3.18 – Network topologies for connectionless messages transfer
a) Star network
b) Bus network
c) Ring network



3.5.2 Individual techniques

CSMA/CD (carrier sense multiple access/with collision detection). This method applies the probabilistic access on the transmission medium which is not synchronized with other stations. The medium is queried for a short period of time before transmission. If it is free, the station transmits, otherwise a waiting period must pass and then the medium is queried again. A collision can occur if a number of stations have ‘queried’ at the same time and then begun to transmit as soon as the medium is free. CSMA/CD is standardized in IEEE 802.3. The network topology is a bus (Figure 3.18 b)). A typical example of CSMA/CD networks is Ethernet. Ethernet can reach a transmission capacity in excess of 10 Mbit/s. Currently work is being conducted on the standardization of a gigabit Ethernet which should reach a transmission capacity of 1 Gbit/s and will also be applicable for wide-area networks.

Token Ring. The token model is a deterministic media access process with a decentralised control system. A transmission permission (token) is passed on from station to station. A station ready to transmit occupies a free token and sends a message. In this way, a new token is created. In the token ring process, the token circulates on a physical ring. The network topology is represented by Figure 3.18 c). The token transfer is carried out along the physical ring. A typical token ring process is the IBM token ring as described in IEEE 802.5.

Token Bus. With this method, all stations connected on a bus (see Figure 3.18 b)) form a logical ring. The token transfer forward is carried out with the addresses of the connected stations. The addresses of the previous and subsequent stations must be known. An example of a token bus process is described in IEEE 802.4.

FDDI (Fibre Distributed Data Interface). FDDI uses the token bus process in a double ring structure with counter-directional rings constructed of fibre optic connections. The data is transported in packets of variable length. FDDI systems are designed to be error-tolerant and are conceived for a high transmission capacity in a High Speed LAN – (HSLAN) of up to 100 Mbit/s. The access procedure permits synchronous service as well as asynchronous data transmission. In this case, every station is assigned a fixed part of the bandwidth.

DQDB (Distributed Queue Dual Bus). While FDDI, token model, and CSMA/CD were developed for the transmission in local area networks, DQDB is the transmission procedure in MAN (Metropolitan Area Networks). It is described in the standard IEEE 802.6.

For DQDB, the transmission is carried out with a frame structure on a double bus running in opposite directions. Depending on which direction the sink is located which is to receive messages from a station, a transmission is requested on the bus of the opposite direction. If a free slot in the desired transmission direction arrives, then it is occupied. With this process, a distributed wait queue develops at each of the stations. The stations can transmit their information with equal rights and without conflicts depending on the general state of the network.

3.6 Abbreviations

ATM	Asynchronous Transfer Mode
BORSCHT	Battery, Over voltage protection, Ringing, Signalling, Coding, Hybrid, Test
CSMA/CD	Carrier Sense Multiple Access/with Collision Detection)
DQDB	Distributed Queue Dual Bus
DSS1	Digital Subscriber Signaling system No. 1
ETSI	European Telecommunications Standards Institute
FDDI	Fibre Distributed Data Interface).
FIFO	First In First Out (normally relating to buffers)
GSM	Group Special Mobile (ETSI committee on second generation cellular systems)
HSLAN	High Speed Local Area Network
IEEE	Institute of Electrical and Electronic Engineers
ISDN	Intergrated Services Digital Network
LAN	Local Area Network

MAN	Metropolitan Area Networks
NNI	Network Network Interface
OSI	Open Systems Interconnection
PCM	Pulse Code Modulation
QoS	Quality of Service
SMS	Short Message Service
UNI	User Network Interface
USBS	User Signalling Bearer Service
WAN	Wide Area Network

CHAPTER 4

4 New signalling systems and SS No. 7

4.1 Introduction

Signalling is a generic term that includes the syntax and the semantics of signals exchanged between user and network or between network nodes to set-up and release calls, or more generally to control the provision of telecommunication services. These signals were originally rudimentary (change of state and frequency combinations). They have since become more and more elaborate in digital networks, taking the form of messages and information elements and taking advantage of the parallel developments of data communication technology (e.g. the OSI model).

The evolution of telecommunication networks towards the IDN (Integrated Digital Network) followed by the N-ISDN (Narrowband Integrated Services Digital Network) saw the emergence of common channel signalling systems in the network (Signalling System No. 7 between network nodes) and in the subscriber loop (Digital Subscriber Signalling system No. 1 between customer premises equipment and the network).

These systems have over the last 20 years really become the nervous systems of modern telecommunication networks, and have met the goals set by the initial designers in the 70s, – reliability and flexibility. They have evolved from the provision of signalling capabilities for basic POTS services, to the provision of a complex “toolbox”, a family of functions and protocols, needed by the network operators to build a variety of services for their customers (POTS, ISDN, cellular, broadband...) and to optimize their network operation (e.g. the Intelligent Network concept). Current developments are tackling the convergence or the interworking with the IP world.

The following sections aim to give an overview of the current developments.

4.2 Signalling system No. 7 (SS7)

Signalling System No. 7 was the second common channel signalling system recommended by CCITT in 1980, after 8 years of study. It integrated in the telephony environment, the principles and techniques of data communication networks, developed in the framework of the Open System Interconnection model, while taking into account the specific signalling requirements such as high level of security and real-time constraints.

In order to provide flexibility and the ability to evolve, the system is designed in a modular and structured manner, with a clear separation between the basic functions providing the reliable transport of signalling messages (Message Transfer Part) and the application related functions (User Parts). Some User Parts may also share common functions such as the Signalling Connection Control Part (SCCP) and the Transaction Capability Application Part (TCAP). This continuous effort towards more structure and commonality between signalling applications can be easily seen in the successive edition of SS7 Recommendations since 1980. The following sections give an overview of the family of SS7 protocols for narrowband services.

4.2.1 Message Transfer Part (MTP)

4.2.1.1 Introduction

The Message Transfer Part (MTP) provides the functions that enable significant information from the User Part to be passed via the MTP across the Signalling System No. 7 network to the required destination (signalling points). In addition, functions are included in the MTP to enable network and system failures that would affect the transfer of signalling information, to be overcome. MTP procedures ensure that messages originated by a user at a signalling point are delivered to the same user at the destination point indicated by the sender without loss and duplication of messages.

The MTP is split into sub-levels:

1) Physical level (level 1) or signalling data link;

2) Data Link level (level 2) or signalling link;

3) Network level (level 3) or signalling network management.

The functions of each level of the MTP are performed by means of the level protocol between two systems which provides a "level service" to the upper levels (i.e. level 1 Signalling Data Link, level 2 Signalling Link and level 3 Signalling network) as described in Recommendations Q.702, Q.703 and Q.704 respectively.

The overall objectives of the Message Transfer Part are to provide the means for:

a) the reliable transport and delivery of "User Part" signalling information across the SS7 network;

a) the ability to react to system and network failures that affect the transport, and take the necessary action to ensure that the permanence of the service.

4.2.1.2 Description

For more details refer to ITU-T Recommendations Q.702 up to Q.707.

Level 1 (signalling data link)

A signalling data link constitutes of the lowest functional level (level 1) in the Signalling System No. 7 functional hierarchy. It is a bi-directional transmission path for signalling, comprising two data channels operating together in opposite directions at the same data rate.

A digital signalling data link is made up of digital transmission channels and digital switches or their terminating equipment providing an interface to signalling terminals. The digital transmission channels may be derived from a digital multiplex signal at 1544, 2048 or 8448 kbit/s having a frame structure as defined in Recommendation G.704 [1].

Signalling System No. 7 is able to operate over both terrestrial and satellite transmission links.

Level 2 (signalling link)

The signalling link functions, together with a signalling data link as a bearer, provide a signalling link for reliable transfer of signalling messages between two directly connected signalling points.

The main features are message delimitation, sequence validation, error checking and flow control. When an error occurs on a signalling link, the message (or set of messages) is retransmitted. Two retransmission methods are defined, the basic procedure for terrestrial links where round trip delay is less than 30 ms, and the Preventive Cyclic Retransmission procedure mainly for satellite links or terrestrial links with round trip delay greater than 250 ms. For round trip delays which are between the two boundaries, any method can be used.

A MTP level 2 message is called a signal unit (SU). There are three kinds of signal units: Fill-In Signal Units (FISUs), Link Status Signal Units (LSSUs), and Message Signal Units (MSUs).

Fill-In Signal Units (FISUs) are transmitted continuously on a signalling link in both directions unless other signal units (MSUs or LSSUs) are present.

FISUs carry basic level 2 information only (e.g., acknowledgement of signal unit receipt by the remote signalling point). As a checksum is calculated for each FISU, signalling link quality is checked continuously by both signalling points at each end of the link.

Link Status Signal Units (LSSUs) carry one or two octets of link status information between signalling points at either end of a link. This information is used to control link alignment and to indicate the status of the link at the remote end.

Message Signal Units (MSUs) carry all user data given by the upper level (MTP level 3).

The differentiation between FISU, LSSU and MSU is made based on the length of the SU. The value of the LI (Length Indicator) field determines the signal unit type:

LI	Message type
0	FISU
1 or 2	LSSU
3 .. 63	MSU

The message length is limited up to 279 octets.

For more details refer to ITU-T Recommendation Q.703

Level 3 (Signalling network management)

Signalling network management is divided into two basic categories:

– Messages handling;

– Network management.

MTP Level 3 is equivalent in function to the *OSI Network Layer*.

Message handling

MTP Level 3 routes messages based on the routing label of message signal units. The routing label is comprised of the destination point code (DPC), originating point code (OPC) and signalling link selection (SLS) fields. The destination and the originating point code are uniquely identified by numeric addresses in the SS7 network. At the destination, the MTP distributes the message to the appropriate MTP user identified in the message (SIO).

Network management

Groups of procedures are provided to manage failure and recovery of elements in the network (link, linkset, and node). Groups are:

– Link management;

– Route management;

– Traffic management.

Whenever a change in the status of a signalling link, route, or node occurs, the different procedures in the sets are activated as appropriate.

Link management

Linkset control	offers management at linkset level.
Signalling link activity control	to supervise all actions made at link level.
Signalling link activation control	used for first activation and link introduction in the network management.
Signalling link restoration	used to restore a failed link after traffic management procedure completion.
Signalling link deactivation	to stop traffic on a link and to delete the link from the network management.

Route management

Transfer prohibited control	Transfer Prohibited (TFP) is used by a Signalling Transfer Point to inform all adjacent nodes that a particular individual destination is no longer reachable through it.
Signalling route set test control	is used to check the routes status from a remote STP knowledge.
Transfer allowed control	Transfer Allowed (TFA) is used by a Signalling Transfer Point to inform all adjacent nodes that a particular individual destination is now reachable through it.
Transfer controlled control	is used to inform adjacent node that path to reach a destination is currently congested.

Traffic management

Changeover control	performs traffic diversion from a currently failed signalling link towards one or more available signalling link.
Changeback control	performs traffic diversion from one or more backup signalling links towards an available signalling link which had previously failed.
Link availability control	performs the synchronization of different events that have impact on routing tables at a link basis.
Forced rerouting control	performs traffic diversion from a currently failed signalling route towards one or more available signalling routes.
Controlled rerouting control	performs traffic diversion from one or more backup signalling routes towards an available signalling route which had previously failed.
Signalling point restart	performs smooth introduction of the node in the SS7 network when restarting. It is also used when an adjacent node is restarting.
Signalling traffic flow control	performs the management of a congestion situation detected either locally or remotely.
Signalling routing control	performs the synchronization of different events that have impact on routing tables at a destination basis.

For more details refer to ITU-T Recommendation Q.704.

4.2.1.3 Future developments

Since 1996, ITU-T Q.704 Recommendation provides, as a national option, a definition of a High Speed Signalling link that allows a 2 Mbit/s signalling link based on non-channelized PCM. The used of such a link is for countries where ATM transport is not yet in operation. For those countries where ATM transport is available a solution using MTP3, ATM in place of MTP levels 1 and 2 is preferable.

4.2.2 Signalling Connection Control Part (SCCP)

4.2.2.1 Introduction

The Signalling Connection Control Part (SCCP) provides, above the MTP network or networks, connectionless, and connection-oriented services to transfer circuit related and non-circuit related signalling information or other types of information between nodes for management and maintenance purposes. The combination of the MTP and the SCCP is called the "network service part" (NSP). The Network Service Part (NSP) of SS7 provides both connectionless and connection-oriented protocol. The NSP of SS7 should be seen as the network layer of the OSI Reference Model.

The SCCP provides the means to:

– control logical signalling connections in a SS7 network;

– transfer Signalling Data Units across the SS7 network with or without the use of logical signalling connections.

The protocol used by the SCCP to provide network services is subdivided into four protocol classes, defined as follows:

– Class 0: Basic connectionless class;

– Class 1: Sequenced connectionless class;

– Class 2: Basic connection-oriented class;

– Class 3: Flow control connection-oriented class.

4.2.2.2 Description

SCCP provides a routing function which allows signalling messages to be routed to a signalling point based on, for example, dialled digits. This capability involves a translation function which translates the global title (e.g. dialled digits) into a signalling point code and possibly a sub-system number. In the routing function, congestion control measures are provided to reduce traffic in the event of MTP congestion, SCCP congestion and SCCP node congestion.

SCCP also provides a management function, which controls the availability of the “sub-systems”, and broadcasts this information to other nodes in the network, which have a need to know the status of the “sub-system”. An SCCP sub-system is an SCCP User.

The SCCP intends to provide compatibility functions to allow transparent and optimal use of the different nature of MTP networks (networks using narrowband links or broadband links such ATM). Unfortunately in the current version this function is provided only to connectionless services. Further study is needed to extend this function on connection oriented services.

For more details refer to ITU-T Recommendations Q.711 up to Q.716.

Routing function

Addressing is the first essential aspect of communication that allows establishing an association between two SCCP users. SCCP allows users to use logical addressing of the remote entity. In such a case, Global Title Translation derives an SS7 Network Address (e.g. the physical address of this remote user). If the association between the logical and the final physical address is not locally possible, the SCCP delivers the message to the next SCCP node to provide the association. This propagation is done until a translation in a SCCP node along the path delivers the final association between this address and a local SCCP user.

The various addressing information elements used by SCCP routing function (SCRC) are:

1) **Point Code (PC)** – This uniquely identifies a node in an MTP SS7 network. This PC identities the next SCCP node.
2) **Global Title (GT)** – This is an address used by the SCCP, comprising dialled digits or another form of address, that will not be recognized by the SS7 Network Layer. Therefore, translation of this information to an SS7 Network Address is necessary.
3) **Sub-System Number (SSN)** – This identifies a sub-system accessed via the SCCP within a node and maybe a User Part (e.g. SCCP Management or an Application Entity containing the TCAP layer).

According to the priority of the message and to the congestion level of the destination (next SCCP node) derived from the GT, the message is sent or discarded.

According to network capabilities all along the path, segmentation is performed or not.

Connectionless services

The connectionless protocol classes provide those capabilities that are necessary to transfer one user message in the “data” field of an SCCP message.

When one connectionless message is not sufficient to convey the user data making use of MTP services provided by a narrowband MTP network that supports a maximum MTP message length of 272 octets including the MTP routing label, a segmenting/reassembly function for protocol classes 0 and 1 is provided. In this case, the SCCP at the originating node or in a relay node provides segmentation of the information into multiple segments prior to transfer in the "data" field of SCCP messages. At the destination node, the segments are reassembled before delivery to the user.

If it is certain that only MTP services supporting long messages (according to Recommendation Q.2210) are used in the network, then no segmentation of the information is needed.

Protocol class 0

User data, passed by higher layers to the SCCP in the originating node, is delivered by the SCCP to higher layers in the destination node. They are transferred independently of each other. Therefore, they may be delivered to the SCCP user out-of-sequence. Thus, this protocol class corresponds to a pure connectionless network service.

Protocol class 1

In protocol class 1, the features of class 0 are complemented by an additional feature (i.e. the sequence control parameter contained in the request primitive) which allows the higher layer to indicate to the SCCP that a given stream of User data shall be delivered in-sequence. The originator chooses the Signalling Link Selection (SLS) parameter used for routing in the MTP SCCP based on the value of the sequence control parameter. The SLS shall be identical for a stream of User data with the same sequence control parameter. The MTP then encodes the Signalling Link Selection (SLS) field in the routing label of MTP messages relating to such User data, so that their sequence is, under normal conditions, maintained by the MTP and SCCP. With the above constraints, the SCCP and MTP together ensure in-sequence delivery to the user. Thus, this protocol class corresponds to an enhanced connectionless service, where an additional in-sequence delivery feature is included.

Connection oriented services

The connection-oriented protocol classes (protocol classes 2 and 3) provide the means to set up signalling connections in order to exchange a number of related user messages. The connection-oriented protocol classes also provide a segmenting and reassembling capability. If a user message is longer than 255 octets, it is split into multiple segments at the originating node, prior to transfer in the “data” field of the SCCP messages. Each segment is less than or equal to 255 octets. At the destination node, the user message is reassembled before delivery to the user.

Protocol class 2

In protocol class 2, bi-directional transfer of user data between the user of the SCCP in the originating node and the user of the SCCP in the destination node is performed by setting up a temporary or permanent signalling connection consisting of one or more connection sections. A number of signalling connections may be multiplexed onto the same signalling relation. Each signalling connection in such a multiplexed stream is identified by using a pair of reference numbers, referred to as “local reference numbers”. Messages belonging to a given signalling connection shall contain the same value of the SLS field to ensure sequencing as described in sequenced connectionless. Thus, this protocol class corresponds to a simple connection-oriented network service, where SCCP flow control and loss or mis-sequence detection are not provided.

Protocol class 3

In protocol class 3, the features of protocol class 2 are complemented by the inclusion of flow control, with its associated capability of expedited data transfer. Moreover, an additional capability of detection of message loss or mis-sequencing is included for each connection section; in such a circumstance, the signalling connection is reset and a corresponding notification is given by the SCCP to the higher layers.

Management

The goal of SCCP management is to provide procedures to maintain network performance by rerouting or throttling traffic in the event of failure or congestion in the network.

For this purpose, SCCP management procedures rely on:

1) failure, recovery, and congestion information provided in the MTP management indication primitives (MTP-pause indication, MTP-resume indication and MTP-status indication); and

2) sub-system failure and recovery information, and SCCP (SSN = 1) congestion received in SCCP management messages.

SCCP management maintains the status of remote SCCP nodes and the status of remote or local subsystems. It informs local users through local broadcast and remote SCCP nodes with either broadcast or response to messages received.

SCCP management co-operates with the SCCP routing control (including translation function) to stop traffic to inaccessible destinations and to provide rerouting of traffic through alternate routing or through selection of alternate remote subsystems.

SCCP management information is transferred using the SCCP connectionless service.

4.2.2.3 Future developments

No major enhancement is foreseen.

4.2.3 Transaction Capabilities (TC)

4.2.3.1 Introduction

The Transaction Capabilities (TC) of Signalling System No. 7 provide support for interactive signalling applications that are not associated to a bearer. Typical applications are interactions between a switching centre and an external database or between two databases. At present, the major users can be found in the area of Intelligent Networks (i.e. INAP) and mobile networks (i.e. MAP). TC may be viewed as a particular realization of the Remote Operations paradigm defined in Recommendation X.880 (ROS). Interactions between TC-Users are modelled by operations. An operation is invoked by an (originating) entity, the other (destination) entity attempts to execute the operation and possibly returns the outcome of this attempt.

4.2.3.2 Description

TC is best understood as a package that provides a Remote Procedure Call (RPC) mechanism over SS7 networks. In OSI terms, it can be compared to the collection formed by the Remote Operations Service Element (ROSE – See Recommendation X.881 and X.882), Association Control Service Element (ACSE – See Recommendation X.217 and X.227) and in some way the Presentation Layer (See Recommendation X.216 and X.226).

TC is modelled as being composed of two sub-layers:

a) The component sub-layer (CSL) which provides the actual remote operation service and allows the TC-User to control the association (if any) over which the requests and replies are exchanged.

b) The Transaction sub-layer (TSL) which provides (if required) an end-to-end connection service between two TC entities.

Transaction Sub-Layer: The Transaction sub-layer directly interfaces with the Signalling Connection Control Part (SCCP) which provides the SS7 connectionless network service. In some ways, it replaces some missing transport layer functionality. Similar to the transport layer, the TSL is used either in a connection-oriented mode, which implies the establishment of a transaction, or in a connectionless mode. The TSL uses the following messages to transport the protocol data units exchanged between two component sub-layers:

– The Begin message is used to open a transaction.

– The Continue message is used to transfer information during an established transaction.

– The End message is used to close a transaction.

– The Abort message is used to refuse the establishment of a transaction, or to abandon an established transaction in case of protocol errors.

Each message can carry one or more protocol data units generated by the component sub-layer.

Component Sub-Layer: The component sub-layer is further divided into two functional blocks:

a) The component handling (CHA) block which provides the core remote operations service;

b) The dialogue handling (DHA) block that provides a simple association control facility.

The protocol data units that are generated or interpreted by the CHA are called “components”. These are:

– The Invoke component is used to request a remote user to perform an action (i.e. to execute an operation).

– The Return Result Not Last and Return Result Last component are used to signal the successful completion of an operation.

– The Return Error component is used to signal the unsuccessful completion of an operation.

– The Return Reject component is used to signal a protocol error.

These PDUs are completely aligned with those defined in Recommendation X.880, with one extension. This addition is a specific PDU, Return Result Not Last, whose sole purpose is to carry segments of the result of a successfully completed operation whose length exceeds the maximum length of connectionless SCCP user-data.

The DHA services are used to control of the establishment and release of a dialogue (association). Controlling the dialogue establishment includes the ability to negotiate an application context. The DHA also enables the TC-User to control the concatenation of one or more PDUs in a single TSL message. The protocol data units of the DHA are a subset of those defined in Recommendation X.217 for the Association Control Service Element (ACSE).

4.2.3.3 Future developments

No further changes to TC Recommendations are expected. Additional common application layer elements (e .g.; for security management) may be defined for use in combination with TC. However, it is assumed that they will not require any enhancements to TC.

4.2.3.4 Road map

The following Recommendations currently define the Transaction Capabilities of Signalling System No. 7:

– Recommendation Q.771 (06/97): Transaction Capabilities – Functional description of transaction capabilities.

– Recommendation Q.772 (06/97): Transaction Capabilities – Transaction capabilities information element definitions.

– Recommendation Q.773 (06/97): Transaction Capabilities – Transaction capabilities formats and encoding.

– Recommendation Q.774 (06/97): Transaction Capabilities – Transaction capabilities procedures.

– Recommendation Q.775 (06/97): Transaction Capabilities – Guidelines for using transaction capabilities.

4.2.4 ISDN User Part (ISUP)

4.2.4.1 Introduction

Since the ’80s, telephone calls have changed from simple calls, consisting of setting-up a circuit between two users, to more and more complicated calls with new services for users as well as new features in the networks. This made it necessary to define a new signalling protocol capable of carrying a lot of information, in the forward direction and also in the backward direction, during the call set-up as well as during the call itself. The ISDN User Part (ISUP) was defined to support the ISDN, that is allowing ISDN subscribers to have their services taken into account, but also offering new services to the analogue subscribers and covering new requirements coming from the Intelligent Network (IN) concept.

4.2.4.2 Description

4.2.4.2.1 Call identification and routing

Since the signalling information is carried on dedicated and circuit independent signalling links, each message contains information allowing every switch involved in a call to bind all messages among themselves within the call, and also to the circuit seized for the call. Each message contains a routing label as defined in the MTP recommendation (originating point code : point code of the sending switch;

destination point code: point code of the adjacent switch) and the CIC (Circuit Identification Code: a unique circuit identity between two adjacent switches). The IAM (Initial Address Message), first message of the call set-up contains the Called party address that is the E.164 address of the called user, used to route the call. By this way, each message, whatever its direction, is recognized as related to a unique call by any switch on the call path.

4.2.4.2.2 ISUP syntax

The ISUP signalling consists of messages. Each ISUP message is contained in the information field of an MTP message. As was stated above, each message contains the routing label followed by the CIC, by the message type code and by the application information itself. This application information is composed of mandatory parameters of fixed length, pointers to each mandatory parameters of variable length and to the optional part, mandatory parameters of variable length, optional parameters of variable length.

4.2.4.2.3 ISUP features

The purpose of ISUP is to support basic bearer services and supplementary services for applications in an integrated services digital network

4.2.4.2.3.1 Basic services

The ISUP usually supports 3 bearer services: speech, 3.1 kHz audio and 64 kbit/s, but it is capable of multi-rate and N × 64 kbit/s. This bearer service is initiated by the originating local exchange in case of analogue calling user (usually 3.1 kHz) or by the ISDN calling user in the DSS1 Set-up message.

For the routing aspect, besides the called party number, the ISUP allows an ISUP signalling to be required (preferred or mandatory) and to indicate in the backward direction if the ISUP has been used.

Moreover, ISUP messages convey information elements provided by ISDN users.

The ISUP also provides procedures for the circuit supervision (blocking/unblocking, reset) and for echo control.

4.2.4.2.3.2 Simple segmentation

The simple segmentation procedure uses the segmentation message to convey an additional segment for the transfer of certain messages whose contents are longer than 272 octets (MTP limit) but no longer than 544 octets.

4.2.4.2.3.3 Compatibility

A main issue in the ISUP protocol is that 2 switches can be interconnected without having exactly the same implementation of the ISUP. Before ISUP'92, when a switch received an unrecognized message or parameter, the unrecognized information had to be discarded. In ISUP'92, an enhancement was introduced such that each new message or parameter is accompanied by compatibility information indicating to switches that do not recognize the message or the parameter what they have to do (discard, pass-on, release call, ...). This allows a network to be upgraded in certain exchanges (i.e. local exchanges) without having to upgrade other exchanges (i.e. transit exchanges).

4.2.4.2.3.4 Application transport mechanism (APM)

In the ISUP'2000, the support of APM was introduced to convey embedded information related to specific applications such as VPN (Virtual Private Network). APM is a mechanism supported by the ISUP via a specific parameter carried by call handling messages, or by a specific message (part of the ISUP) when call handling messages are not available at the moment APM information is to be sent.

4.2.4.2.3.5 Supplementary services

The ISUP supports several services. This support is based upon the ISUP itself or is combined with generic procedures taking into account the SCCP level. For some supplementary services, the ISUP conveys information that has to be handled by switches. For other ones, the ISUP is only a way of carrying notification information, in particular when a supplementary service is locally provided, e.g. on a DSS1 access.

For each ISUP "version", the supplementary services supported by the ISUP are listed. Among all the supplementary services supported, here are some examples:

– CLIP/CLIR (Calling Line Identification Presentation/Restriction): the called party number is transmitted in the Initial address message to be displayed at the destination access.

– CUG (Closed User Group): the CUG information is transmitted by the Initial address message to be analysed by the CUG application in switches that are concerned.

– UUS (User-to-User Signalling): this ISDN service consists of exchange of information between ISDN users.

– CFU, CFNR, CFB (Call Forwarding Unconditional, No Reply, Busy): the information related to the type of call forwarding and to the called identities is conveyed for handling (e.g. charging purposes) or to be displayed either in local switches or on DSS1 accesses.

– CCBS (Completion of Calls to Busy Subscriber): this service is initiated by the ISUP Release message indicating to the originating local exchange that the busy called user can be called again when he becomes free. The rest of the procedure is supported by SCCP.

4.2.4.3 Future developments

The ISUP protocol is being enhanced to take into account new requirements. For example, in the ISUP'2000, number portability has been considered, that is new procedures such as redirection procedure and pivot routing and new information have been introduced. Other requirements come from IN. Indeed, IN services require more and more information exchanged between IN nodes or between IN nodes and switches or users.

Another evolution is the use of the ISUP to support new bearers such as ATM or IP. To allow a fast ATM/IP backbone networks deployment, a study has started to derive a new ISUP, based on the N-ISUP, capable of call handling independent of the bearer. The result will be a new ISUP protocol, interworking with the N-ISUP.

4.2.4.4 Road-map (Q.761 to Q.764 and Q.730 series)

2000: ISUP'2000 (APM, VPN, number portability, IN CS2, new supplementary services)

1997: ISUP'97 (new procedures, IN CS1, new supplementary services)

1993: ISUP'92 White Book (segmentation, compatibility, new supplementary services)

1991: ISUP Q.767 (subset of ISUP for international interface)

1988: ISUP Blue Book

4.2.5 Intelligent Network Application Part (INAP)

4.2.5.1 Introduction

A broad range of services is now available in modern telecommunications. Market liberalization makes it possible to offer an increased choice of features to customers while boosting revenue streams for operators and services providers.

To compete successfully and make the most of new opportunities, network operators must take a fresh look at network infrastructure, costs and flexibility, and tailor services to changing customer expectations.

Applying the Intelligent Network solution is an answer to this new challenge, applicable for all networks. The main benefit of the IN is the ability:

– to ease the introduction of new services without impacting the underlying network structure;
– to provide service customisation according to individual needs;
– to enable smooth upwards evolution;
– to improve sources of revenue, and to provide a fast return on investment;
– to reduce the cost of network service operations;
– to establish vendor independence;
– to create open interfaces.

4.2.5.2 Description

Intelligent network is an architectural concept which is intended to be applicable to fixed, mobile, Internet, data and other networks as well as narrow-band, data, or multimedia communications.

A phased standardization process has been initialised and is now on the way to deliver the Capability Set 3 Recommendations. In order to ensure a smooth evolution towards the target, each IN Capability Set of Recommendations allows backward compatibility and is open-ended towards long-term views.

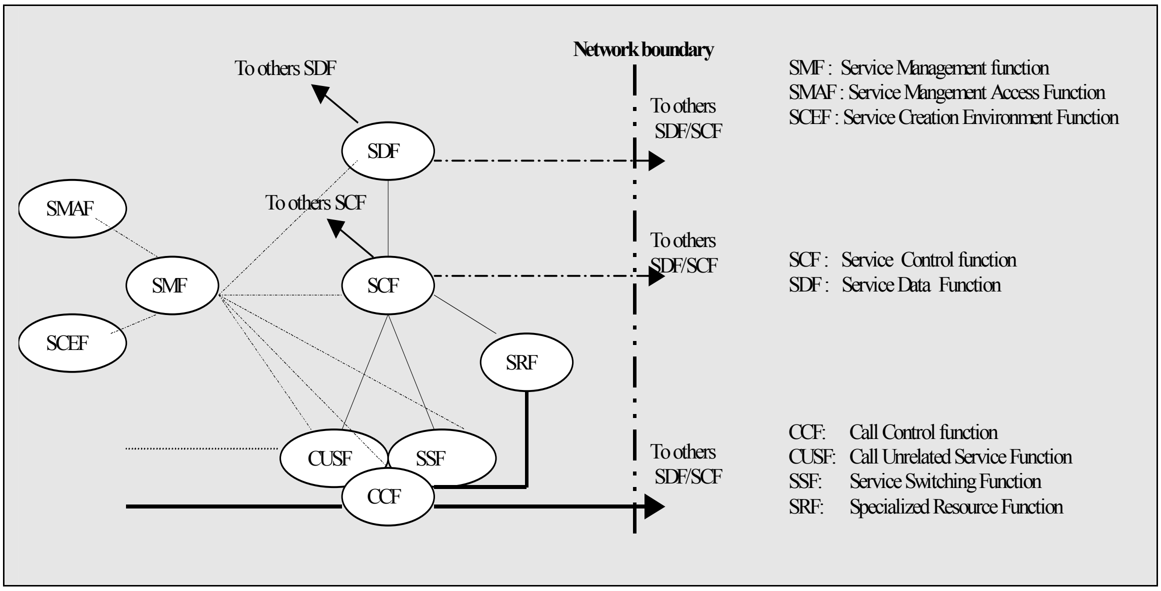
An Intelligent Network is made up of a set of functional entities that fit together as depicted in the figure hereafter. Each of these entities could be supported by an independent physical node but they also could be located in the same platform. Modularity of integration enables an operator to step into IN services at a low level while making it possible to grow to larger IN solutions.

The CCF is the call control function in the network that provides call/connection processing control.

The SSF identifies an IN call, recognizes the requested services, and executes the SCF commands. The SSF is more often supported by a switch e.g. in a fixed or mobile network.

The CUSF is the counterpart of the SSF for calls which do not require the use of a bearer.

Figure 4.1 – Functional entities of an intelligent network



The SCF contains the real time functions of service scenarios, monitors events through the SSF, requests voice prompt and receives user control through specialized resources managed by the SRF function.

The SDF is the database that contains all the service data, e.g. the IN service user profile.

The SMF centralizes all function needs for management, as well as to customize services, configures alarms, and other related functions.

The SMAF provides an interface between the service managers and the SMF.

The SCEF is the software tool set that enables service creation and development.

These functional entities exchange information via standardized interfaces using the ASN1 method for coding. The transport is more often the SS7 TCAP/SCCP system but nothing prevents the use also of TCP/IP technology.

CS1 Recommendations describes the first basic set of IN capabilities applicable for a two party call. These capabilities enable services to provide:

– flexible Triggering;

– flexible Routing;

– flexible Charging;

– flexible User Interaction.

Based on these capabilities, a large range of IN CS1 services are now deployed, e.g. the whole set of prepaid/post-paid card services, Virtual Private Networks services, the 800 numbers services, user mobility as UPT.

CS2 Recommendations enlarge the scope by adding:

- flexible connection command; this enables to supervise IN call with several users involved,
- inter-operator interfacing; refer to SCF-SDF, SDF-SDF, and other functions, see Figure 4-1.
- service for calls without the use of a bearer; e.g. for managing mobility, see CUSF in Figure 4-1.

CS2 services are not yet deployed. They will certainly be much more user oriented than network oriented.

CS3 is under stabilization and could be seen as CS2 refinements, as it does not really add major new features.

More information can be found in Fascicle 2 Chapter 2.3 of this Handbook.

4.2.5.3 Future developments

The future of IN is open, the most important challenges on the table are the integration of the Internet network, the adaptation to Voice Over IP Technology and the mobility with the Virtual Home Environment concept applicable to the new generation of mobile networks in the IMT-2000 family.

4.2.5.4 Road-Map

IN Capability Set 1: Q.1214 to Q.1219 May 1995;

IN Capability Set 2: Q.1224 to Q.1229 September 1997;

IN Capability Set 3: Q.1238.1, Q1238.2, Q.1238.3, Q.1238.4. Q.1231 (planned 2001);

IN Capability Set 4: Q.1241 and Q.1248 Determination planned 2001.

4.3 Digital subscriber signalling system 1 (DSS1)

4.3.1 Introduction

The ISDN protocols extend common channel signalling to terminal equipment on subscriber premises. The D channel of a user-network access line carries the signalling for the channels of the same line. It has the essential purpose of allowing all ISDN services to be offered in an integrated manner. It is intended for use by the many types of subscriber equipment liable to have access to public and private networks.

The D channel protocol is structured into layers:

The physical level (layer 1) of the user-network interface is defined at the S (point to multipoint) or T (point to point) interface in basic (2B + D) or primary (30B + D) rate access.

The link level (layer 2) provides for transmission of frames between the two entities on each side of the interface.

The network level (layer 3) assures control and supervision of calls and supplementary services. It is known as the Digital Subscriber Signalling System Number 1 (DSS1).

4.3.2 Description

4.3.2.1 Call control

DSS1 functions for call control involve message handling and management of the resources assigned to the call, for circuit, packet and connectionless modes.

4.3.2.1.1 Protocol design

Communication between the user and the network for call control purposes is by means of variable length messages. DSS1 provides a procedure so that the messages that are longer than the length of frames that the data link layer can support may be partitioned into several segments.

Several types of messages have been specified, each of them determining an action on the user or network side. For instance:

– SETUP for a call set up request.
– CALL PROCEEDING, as a response to the SETUP, means that call information is complete and the call is being set up.
– ALERTING indicates that the called user is being alerted.
– CONNECT sent at the called user response
– DISCONNECT, RELEASE and RELEASE COMPLETE, used for the release of the call.

Each message contains information elements, usually of variable length, that define the type of request, the resources to be reserved, addressing features or the result of a processing operation. For instance:

– "Bearer capability" indicates the requested bearer service to be provided by the network (Circuit mode / Packet mode, Speech / 64 kbits/sec., etc.)
– "Channel identification" indicates the B or D channel to be used for the call.
– "High layer compatibility" provides a means which should be used by the remote user for compatibility checking.
– "Calling party number" and "Called party number" provides addressing information.

4.3.2.1.2 Control of circuit-switched calls

Each call is provided with a call reference, chosen by the initiating side. The B-channel(s) to be used is negotiated, at the set-up of the call, between the user and the network.

Service indications and addresses are provided by the calling terminal. They determine the functions to be performed in the network and allow the right terminal(s) to be selected on the called subscriber's premises.

4.3.2.1.3 Control of packet-switched calls

ISDN users may have access to packet switched services supplied by a public switched packet data network (PSPDN) or by a Packet Handling function within the ISDN.

When transfer of information is through a B-channel, the D-channel procedure for setting up the connection is the same as for a circuit-switched call. The call request packet is sent when the physical link has been established.

The D-channel can also be used for the transfer of packets.

4.3.2.2 Control of supplementary services

Three methods are specified for the control of supplementary services: the stimulus procedure, the feature key management procedure, and the functional procedure.

In the stimulus protocol, the request is generated by the user by pressing a function key or digit keys. It is not interpreted by the terminal which inserts it in the "keypad" information element(s) of SETUP or INFORMATION message(s) and ignores the type of the request.

In the user to network direction, information is provided in the "Display" information element and displayed on the terminal without interpretation.

Only the network analyses the information to determine the necessary actions, which does not need any standardization of the supplementary services on the user-network interface. The stimulus procedure simplifies terminal design and facilitates the introduction of new supplementary services.

The feature key management procedure is also a stimulus procedure in the sense that it does not require knowledge of the supplementary service by the user's terminal equipment.

It is based on the use of two information elements. The "Feature activation" information element may be included in the SETUP and the INFORMATION messages in the user-to-network direction. The "feature indication" information element may be included in basic call control messages in the network-to-user direction.

The supplementary service associated with the feature identifier is service provider dependant and must be coordinated between the user and the service provider at subscription time.

In the functional procedure, the terminal analyses the user-input information and inserts the request in a dedicated information element or in the "Facility" information element common to several supplementary services and inserted in call control or FACILITY messages.

Both the terminal and the network analyse the information, so the functional procedure needs to be closely defined for every supplementary service. The functional protocol allows for the design of powerful supplementary services.

4.3.2.3 Future developments

New supplementary services are continuously defined in the functional procedure.

The DSS1 has also recently been extended for the handling of calls between users in a corporate telecommunication network.

The DSS1 protocol is being enhanced to take into account new requirements linked to the handling of supplementary services by IN servers, transparently to the originating exchange.

4.4 Broadband signalling

4.4.1 Low layers supporting B-ISDN signalling

UNI

The transfer of the DSS 2 signalling information at the UNI is made by using the protocol stack as follows:

DSS 2
SAAL
ATM (I.361)
PH (I.432)

The PH and ATM layers are not specific to signalling and are the same for both user plane and control plane (UNI and NNI). The physical layer can be any one of the transmission systems capable of carrying ATM cells. The ATM layer allows multiplexing of signalling information with the user plane information on the same physical layer. As per VPI, VCI = 5 is reserved exclusively for the transport of signalling information. At the UNI, the VPI used depends on the UNI access configuration. By default, VPI = 0 is used.

NNI

At the NNI, either the ATM network, or, as a national option, the Signalling System No. 7 network can be used for signalling information transfer. The protocol stack used when an ATM network is used is:

B-ISUP
B-MTP 3 (Q.2210)
SAAL
ATM (I.361)
PH (I.432)

B-MTP 3 is an evolution of the N-ISDN MTP 3 which includes changes to be used above SAAL. These changes are:

– SAAL/B-MTP 3 primitives,

– New code values to identify B-ISUP (e.g. new service indicator),

– Maximum size of SDU is changed from 272 octets to 4091 octets,

– Adaptation of the changeover procedure.

As a national option, when the Signalling System No. 7 network is used, the protocol stack is the following:

B-ISUP
MTP 3 (Q.704)
MTP 2 (Q.703)
MTP 1 (Q.702)

In this case, it should be noted that the segmentation function defined in B-ISUP is to be used.

SAAL

The ATM Adaptation Layer is defined as enhancing the services provided by the ATM layer to support the functions required by the higher layer. SAAL is a particular type of AAL used for the support of signalling. As shown in the previous figures, the SAAL is then used at both sides, UNI and NNI.

The SAAL is a combination of two sub-layers, a common part and a service specific part. The SAAL makes use of the common part and service specific part of AAL type 5 as defined in I.363.5. For SAAL, the service specific functions are performed by a combination of the Service Specific Connection Oriented Protocol (SSCOP) and two types of Service Specific Coordination Function (SSCF), one to map the particular requirements of the UNI (DSS 2 layer 3 protocol) to the SSCOP services, and one to map the particular requirements of NNI to the SSCOP services.

The SSCOP is a protocol which provides mechanisms for the establishment and release of connections and the reliable exchange of information. The SSCFs are not protocols, they only provide mapping between the SSCOP and either DSS 2 or B-MTP 3 + B-ISUP.

ATM forum annex

For both sides, UNI and NNI, the ATM Forum only uses the ATM lower layers for signalling information transfer. The ATM Forum protocol stack is as follows:

UNI 4.0 or PNNI 1.0
SAAL
ATM (Rec.I361)
PH (Rec. I.432)

The UNI 4.0 and PNNI 1.0 are both based on the ITU-T DSS 2 standard. Therefore, the SAAL used at UNI and NNI are the same, the one defined by ITU but making use of only one Service Specific Coordination Function (SSCF), the one defined by ITU-T for UNI (i.e. refers to Recommendation Q.2130).

Relevant ITU-T Recommendations

- Q.2010 (1995): Broadband integrated services digital network overview – CS 1;
- Q.2100 (1994): SAAL Overview description;
- Q.2110 (1994): B-ISDN AAL – Service Specific Connection Oriented Protocol (SSCOP);
- Q.2130 (1994): BISDN SAAL – Service specific coordination function for support of signalling at the user / network interface (SSCF at UNI);
- Q.2140 (1995): BISDN AAL – Service specific coordination function for signalling at the network node interface (SSCF at NNI);
- Q.2210 (1996): MTP 3 functions and messages using the services of the ITU-T Recommendation Q.2140.

4.4.2 Broadband ISDN User Part (B-ISUP)

4.4.2.1 Introduction

This section is an overview of the NNI interface for public ATM networks specified by the ITU-T, namely B-ISUP (Broadband ISDN User Part). This description does not intend to be exhaustive, but especially focuses on features which show ATM networks to advantage, such as traffic and quality of service capabilities, and handling of traffic parameters during the active phase of calls. Finally, a complete listing of B-ISUP Recommendations is provided, as well as a short description of the signalling systems specified by the ATM Forum.

4.4.2.2 B-ISUP signalling protocols

B-ISUP signalling protocols are specified in the ITU-T Q.27xx series of Recommendations. This paragraph mainly covers capabilities contained in Q.2761-4 set of Recommendations. A complete listing of B-ISUP Recommendations is given in Annex A.

4.4.2.2.1 Point-to-point Switched Virtual Channels

The handling (establishment and clearing) of point-to-point ATM switched Virtual Channel connections is described in Recommendations Q.2761-4. The last version of these Recommendations, namely B-ISUP 2000 (07/99), contain the following capabilities:

- support of Quality of Service, as specified in I.356;
- support of Individual Quality of Service parameters;
- support of ATCs as specified in I.371;
- support of negotiation during connection set-up;
- modification procedures;
- support of Frame Relay;
- support of routing on E.164 and AESA address formats;
- crankback procedure;
- negotiation of AAL parameters.

Two other kinds of virtual connections are also defined in B-ISUP:

- Soft Permanent Virtual Channel connection (Soft PVCC);
- Soft Permanent Virtual Path Connection (Soft PVPC).

These capabilities allow re-establishment of either a PVCC or a PVPC using the basic call procedure mentioned above if case of failure. This is specified in the ITU-T recommendation Q.2767.1.

4.4.2.2.2 Signalling of the ATM Transfer Capabilities and associated traffic parameters

End-systems connected to ATM networks will generate a wide diversity of traffic behaviours. Therefore an ATM virtual channel connection has to be set up with the ATM Transfer Capability (ATC) which fits the concerned end-system traffic behaviour.

The ATCs supported in B-ISUP and defined in ITU-T Recommendation I.371, are: Deterministic Bit Rate (DBR), Available Bit Rate (ABR), ATM Block Transfer (ABT DT/IT), Statistical Bit Rate (SBR1, SBR2, and SBR3).

These ATCs are signalled in the Broadband Bearer Capability.

Each ATC is described by a set of traffic parameters:

– DBR: PCR (Peak Cell Rate);

– ABR: PCR, MCR (Minimum Cell Rate), ECR (Explicit Cell Rate), CI (Congestion Indication), NI (No-increase Indication);

– ABT DT/IT: PCR, SCR (Sustainable Cell Rate), BCR (Block Cell Rate);

– SBR: PCR, SCR and MBS (Maximum Burst Size).

These traffic parameters are signalled in the following B-ISUP parameters: ATM Cell Rate, Additional ATM Cell Rate, ABR parameters, minimum ATM Cell Rate.

A successful point-to-point connection set up means that a Virtual Channel connection (VC) with the requested traffic characteristics has been allocated between the Calling and Called users across the ATM network. The compliance of the traffic generated by the source is then checked by the UPC/NPC (Usage Parameter Control/Network Parameter Control), performing the GCRA algorithm.

The Cell Delay Variation Tolerance (CDVT) which is used by GCRA algorithm can also be indicated via signalling. This enables change of GCRA parameters on a per call, and per Virtual Channel connection, basis.

4.4.2.3 Negotiation of traffic parameters

The negotiation procedure provides to the user means to set up a virtual channel connection, for which two sets of traffic parameters can be specified. Two kind of negotiations are offered to the user:

1) based on a maximum and a minimum cell rate specified by the user in its set up request, the network allocates to the connection any possible cell rate allowed by the Connection Admission Control (CAC) function, between these extreme cell rates;

2) either the cell rate specified is the original cell rate or the alternative cell rate is allocated to the connection.

This procedure can be applied, in the forward and backward directions, to the following traffic parameters: Peak Cell Rate, Sustainable Cell Rate, and Maximum Burst Size.

4.4.2.4 Modification of traffic parameters

During the active phase of a call, the calling user, as the owner of the virtual channel connection, has the ability to modify the traffic related resources attributed to that connection.

The traffic parameters which can be modified are: Peak Cell Rate, Sustainable Cell Rate, and Maximum Burst Size.

The modification procedure offers two different options:

1) simple modification: the request issued by the user indicates only one possible value to be considered for modification of the PCR of the virtual connection;

2) modification combined with negotiation: the user is allowed to specify either a maximum and a minimum cell rate value, or an alternative cell rate value. In the former case, the traffic characteristics of the active connection are modified and set to any possible value (allowed by the CAC function) between the minimum and maximum cell rate. In the latter case, the traffic characteristics are modified using the alternative cell rate value.

The combined procedure allows modification of a wider set of traffic parameters, namely Peak Cell Rate, Sustainable Cell Rate, and Maximum Burst Size. Note that ABR and ABT also have their own specific modification capabilities.

4.4.2.5 QoS Class indication and support of individual QoS parameters

As ATM networks aim at transporting information flows with different quality of service classes, a user who initiates an ATM VC connection request must have the ability to indicate his own quality of service requirements. B-ISUP provides such a mechanism, in compliance with the definition of quality of service specified in ITU-T Recommendation I.356.

A user setting up a connection, signals a specific quality of service class, among a set of authorized classes. This class is relevant to routing and resource allocations in the network. If the connection is successfully established, this means that a quality of service corresponding to the indicated class is guaranteed end-to-end by the network.

B-ISUP supports the four QoS classes recommended in I.356 (1995). A QoS class is an aggregate of performance parameters: Cell Delay Variation (CDV), Cell Transfer Delay (CTD), and Cell Loss Ratio (CLR). A particular QoS class is a combination of bounds on these performance parameter values.

Moreover, B-ISUP also supports signalling of the performance parameters individually. This feature is provided for interworking with private ATM networks. Therefore, such parameters are not relevant either to routing, or to resource allocation, and are updated by each B-ISUP node across the network, and delivered to the destination side.

4.4.2.6 Point-to-multipoint SVCs

B-ISUP also specifies signalling procedures for the set-up of unidirectional point-to-multipoint virtual channel connections. Such a connection is made up of a root party, and several leaf parties.

The root party which sets up such a connection, initiates a set-up request with a Broadband Bearer Capability indicating a point-to-multipoint connection, towards a leaf party. The information such as ATM Cell Rate and Broadband Bearer Capability, is stored by the root party. The addition of any subsequent leaf party is performed by the root, upon request from the leaf parties or the root itself.

During the active phase of the point-to-multipoint connection, the dropping of a leaf can be initiated either by the leaf itself or the root. A common procedure is used.

4.4.2.7 Supplementary services

The supplementary services supported in B-ISUP, described in Recommendation Q.2730, are the following:

– User-to-user signalling;

– Calling Line Identification Presentation/Restriction;

– Direct Dialling in;

– Connected Line Identification/Restriction;
– Sub-addressing;
– Multiple Subscriber Number;
– Closed User Group.

As B-ISUP supports two different address formats (E.164 and AESA), services such as Calling Line Identification Presentation/Restriction have also been extended for AESA format.

4.4.2.8 ITU-T Recommendations

Table 2 – ITU-T B-ISUP Recommendations

Signalling function	B-ISUP
Point-to-point call/connection control	Q.2761 (07/99) + Q.2762 (07/99) + Q.2763 (07/99) + Q.2764 (07/99)
Soft PVC Capability	Q.2767.1
Generic functional protocol : core functions	TCAP
Signalling specification for frame relay service	Q.2727 (07/99)
Switched virtual path capability	Q.2766.1 (05/98)
Soft PVC capability	Q.2767.1 (05/98)
Network generated session identifier	Q.2726.3 (07/96)
Generic identifier transport	Q.2726.4 (09/97)
Supplementary services (CLIP, CLIR, MSN, DDI, SUB, COLP, COLR)	Q.2730 (07/99)
Closed user group (CUG)	Q.2735.1 (06/97)
User-to-user signalling (UUS)	Q.2730 (02/95)
Call priority	Q.2726.2 (07/96)
Sustainable cell rate parameter set and tagging option	Q.2723.1 (07/99)
Support of ATM transfer capability	Q.2723.2 (09/97)
Signalling capabilities for the support of ABR	Q.2764 (07/99)
Signalling capabilities for the support of ABT	Q.2723.4 (07/99)
Cell delay variation tolerance indication	Q.2723.5 (07/99)
Support of the SBR2 and SBR3 capabilities	Q.2723.6 (07/99)
Connection characteristics negotiation	Q.2761-4 (07/99)
Peak cell rate modification by the connection owner	Q.2761-4 (07/96)
Modification procedures for sustainable cell rate parameters	Q.2761-4 (07/99)
Modification with negotiation	Q.2761-4 (07/99)
Basic look ahead	Q.2724.1 (07/96)
Support of QoS classes	Q.2761-4 (07/99)
Point-to-multipoint call/connection control	Q.2722.1 (07/96)

n.s.: not specified

4.4.3 Digital Signalling System Number 2 (DSS2)

4.4.3.1 Introduction

This section provides an overview of the Digital Signalling System No. 2 (DSS2) specified by ITU-T Study Group 11. The DSS2 signalling system enables the control of switched ATM connections at the User-to-Network Interface (UNI) of the Broadband Integrated Services Digital Network (see Rec. I.413).

Annex A gives a list of the DSS2 signalling Recommendation in force. Annex B provides a general overview of the differences between the ITU-T DSS2 signalling system and the ATM Forum UNI signalling specifications (UNI 3.1/4.0).

4.4.3.2 DSS2 signalling protocols

DSS2 signalling protocols are specified in the ITU-T Q.29xx series of Recommendations. This clause gives a description of the major capabilities currently supported in DSS2.

4.4.3.2.1 Point-to-point ATM switched virtual channels (Rec. Q.2931)

Recommendation Q.2931 published in 1995 constitutes the basic foundation of the DSS2 signalling system, on top of which extensions are built for the support of additional signalling capabilities such as point-to-multipoint connection control, negotiation/modification of ATM traffic characteristics, as described in the following sub-clauses. This Recommendation is closely based on the equivalent narrowband DSS1 Q.931 Recommendation and has been adapted for the control of ATM virtual channels instead of 64 kbit/s based circuits.

Recommendation Q.2931 specifies the signalling protocol for the establishment, maintaining and clearing of ATM point-to-point Switched Virtual Channels (SVC) at the B-ISDN User-to-Network Interface (UNI). The basic capabilities supported by this Recommendation are:

– support of point-to-point bi-directional connections with symmetric or asymmetric bandwidth requirements;

– support of one and only one connection per call;

– signalling of the ATM traffic characteristics requested for the SVC; This consists of an indication of the I.371 ATM transfer capability (ATC) and the associated ATM traffic parameters. Rec. Q.2931 (1995) enables the establishment of constant bit rate (CBR) ATM bearers with symmetric or asymmetric peak cell rate characteristics. Since the publication of Rec. Q.2931 (1995), new ATM transfer capabilities have been defined and can be signalled as specified in the Q.2961 series of Recommendations (see clause 2.3 for more details);

– signalling of the Quality of Service (QoS) characteristics requested for the SVC; This consists of an indication of the QoS class and optionally of end-to-end transit delay requirements. Rec. Q.2931 (1995) only allows for the signalling of the implicit QoS class associated with the requested ATC. Since the publication of Rec. Q.2931 (1995), explicit signalling of the requested QoS class is supported in DSS2 (see Q.2965.1 in clause 2.6);

– support of E.164 addressing schemes and ATM End System Address (AESA) as specified in ITU-T Recommendation E.191;

– support for the signalling of user application dependent characteristics; This consists of signalling the characteristics of the layers above the ATM layer such as the ATM adaptation layer or the layers above.

Note – Recommendation Q.2933 extends Rec. Q.2931 by defining a B-ISDN frame relay bearer service allowing a single frame relay connection to be carried on an ATM SVC using AAL type 5 and the I.365.1 service specific convergence sub-layer.

4.4.3.2.2 Point-to-point ATM switched virtual paths (Rec. Q.2934)

This Recommendation specifies extensions to Rec. Q.2931 allowing the control of Switched Virtual Paths (SVP).

4.4.3.2.3 Signalling of ATM traffic capabilities (Rec. Q.2961)

This multi-part Recommendation provides extensions to Rec. Q.2931 to enable the signalling of I.371 ATM transfer capabilities such as DBR, SBR, ABR or ABT. In addition, a specific Q.2961 part provides extensions in order to support the (optional) signalling of the associated I.371 CDVT values (Cell Delay Variation Tolerance).

4.4.3.2.4 Negotiation of ATM traffic parameters (Rec. Q.2962)

This signalling function specifies extensions to Rec. Q.2931 that allow a calling user to negotiate with the network and the called user, on the traffic characteristics of a connection during the call/connection establishment phase.

Two exclusive modes of negotiation are defined in Q.2962:

– negotiation between a set of traffic parameter values and a set of alternative ATM traffic parameter values;
– negotiation of individual ATM traffic parameters between their respective requested nominal value and their respective minimal acceptable traffic parameter value.

4.4.3.2.5 Modification of ATM traffic parameters (Rec. Q.2963)

This multi-part Recommendation defines extensions to Rec. Q.2931 that allow the calling user to request a modification of the ATM traffic parameters of an already established point-to-point connection (i.e. during the active phase of the call/connection).

Part 1 is limited to the modification of Peak Cell Rate values, while Part 2 extends the modification to the Sustainable Cell Rate parameter set. Part 3 enhances the previous two parts by allowing negotiation to be performed while modifying traffic parameters.

4.4.3.2.6 Signalling of Quality of Service (QoS) characteristics (Rec. Q.2965)

Recommendation Q.2965.1 amends Rec. Q.2931 and specifies the signalling of the Quality of Service classes as defined in Rec. I.356. Recommendation Q.2965.2 (approved December 1999) provides extensions to Q.2965.1 in order to enable the signalling of individual QoS parameters (i.e. Cell Transfer Delay (CTD), Cell Delay Variation (CDV) and Cell Loss Ratio (CLR).

4.4.3.2.7 Point-to-multipoint Switched Virtual Channels (Rec. Q.2971)

This DSS2 signalling Recommendation specifies the control of point-to-multipoint switched connections which once established allow a source (named “root” in Q.2971) to broadcast information to a root selected number of destinations (called “leaves” in Q.2971). A point-to-multipoint virtual channel connection is a collection of associated ATM virtual channel links that connect two or more endpoints. This capability only supports unidirectional transport from the root to the leaves. Parties can be added and removed during the lifetime of the connection.

This Recommendation is based on extensions to Rec. Q.2931 for the initial establishment of the connection between the source and one leaf.

After this establishment request has progressed to alerting or active state, additional leaves can be added to the connection by “add party” requests from the root allowing the root to widen the broadcast of information. A leaf may be added or dropped from the call at any time while the call is in the active state. A new leaf can be added via the root issuing an ’add party’ request, as described above. A leaf can be dropped from the call by either the root or by the leaf.

Multiple add party requests pending at the same time are allowed (e.g. the root node does not need to wait for a response related to one 'add party' request before issuing the next one). Multiple 'drop party' requests pending at the same time are allowed (i.e. the root does not need to wait for a response related to one 'drop party' request before issuing the next one). Each 'add party' or 'drop party' request is exclusively related to one party. If, as a result of a 'drop party' procedure, there are no leaf parties remaining in the call, the entire call is released.

The root can also release at any time the entire connection and the attached leaves.

4.4.3.2.8 Generic identifier transport (Rec. Q.2941)

This signalling function allows the transport of application dependent identifiers between end-systems when establishing a connection or when releasing a connection. The information is carried transparently by the networks, through which the connection is established. This end-to-end transfer of signalling identifiers is useful in order to correlate at the application level, multiple calls as being part of the same session. This signalling function is used in many end-system distributed applications when making use of ATM signalling such as H.321/H.310, DSMCC, and H.323 over ATM.

4.4.3.2.9 Supplementary services (Q.295x series)

DSS2 supplementary services are defined in the Q.295x series. They support the following signalling capabilities:

– additional addressing capabilities such as Direct Dialling In (DDI), Multiple subscriber number (MSN) or Sub-addressing (SUB);

– calling line identification (CLIP) and the associated restriction service (CLIR);

– connected line identification (COLP) and the associated restriction service (COLR);

– user-to-user signalling type 1 which allows the exchange of information between end-systems during the establishment phase or the release phase (see Q.2957.1);

– closed user group facilities (CUG) as specified in Q.2955.1;

– handling of multiple priority levels during call/connection (i.e. Q.2959).

4.4.3.2.10 Generic functional protocol (Rec. Q.2932.1)

Recommendation Q.2932.1 is an adaptation of the DSS1 Q.932 Recommendation in order to enable similar functions to be applicable in the context of ATM signalling. This protocol enables the transport of remote operations locally across the UNI. Three transport modes of ROSE operations are defined and can be used depending on the signalling and user application requirements. The bearer related transport mode defines the transfer of ROSE operations in relation with existing bearers controlled via ATM signalling. The connection oriented bearer independent (COBI) and the Connectionless bearer independent (CLBI) transport modes allow the transfer of ROSE operations independently of any existing bearer and differs one in the way ROSE operations are carried at the UNI.

4.4.3.3 ITU-T Recommendations

Table 4.2 – ITU-T DSS2 signalling recommendations in force (03/99)

Signalling function	DSS2
Point-to-point call/connection control	Q.2931 (02/95) + Amd.1 (06/97) + Amd.2 (03/99) + Amd.3 (03/99)
Generic functional protocol: core functions	Q.2932.1 (07/96)
Signalling specification for frame relay service	Q.2933 (07/96)
Switched virtual path capability	Q.2934 (05/98)
Generic identifier transport	Q.2941.1 (09/97)
Supplementary services (CLIP, CLIR, MSN, DDI, SUB, COLP, COLR)	Q.2951.x (02/95)
Closed user group (CUG)	Q.2955.1 (06/97)
User-to-user signalling (UUS)	Q.2957.1 (02/95)
Call priority	Q.2959 (07/96)
Sustainable cell rate parameter set and tagging option	Q.2961.1 (10/95)
Support of ATM transfer capability	Q.2961.2 (06/97)
Signalling capabilities for the support of ABR	Q.2961.3 (09/97)
Signalling capabilities for the support of ABT	Q.2961.4 (09/97)
Cell delay variation tolerance indication	Q.2961.5 (03/99)
Support of the SBR2 and SBR3 capabilities	Q.2961.6 (05/98)
Connection characteristics negotiation	Q.2962 (05/98)
Peak cell rate modification by the connection owner	Q.2963.1 (07/96)
Modification procedures for sustainable cell rate parameters	Q.2963.2 (09/97)
Modification with negotiation	Q.2963.3 (05/98)
Basic look ahead	Q.2964.1 (07/96)
Support of QoS classes	Q.2965.1 (03/99)
Point-to-multipoint call/connection control	Q.2971 (10/95)

n.s.: not specified.

4.5 Abbreviations

ABR	Available Bit Rate
ABT	ATM Block Transfer
ACSE	Association Control Service Element
AESA	ATM End System Address
AINI	ATM Inter Network Interface

APM	Application transport mechanism
ASN1	Abstract Syntax Notation No. 1
ATC	ATM Transfer Capability
ATM	Asynchronous Transfer Mode
BCR	Block Cell Rate
B-ISDN	Broadband-Integrated Services Digital Network
B-ISUP	Broadband-ISDN User Part
CAC	Connection Admission Control
CBR	Constant Bit Rate
CCBS	Completion of Calls to Busy Subscriber
CCF	Call Control Function
CDV	Cell Delay Variation
CDVT	Cell Delay Variation Tolerance
CFB	Call Forwarding Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CHA	Component Handling
CI	Congestion Indication
CIC	Circuit Identification Code
CLBI	Connectionless Bearer Independent
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CLR	Cell Loss Ratio
COBI	Connection Oriented Bearer Independent
COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
CS1, 2, 3, 4	Capability Set No. 1, 2, 3, 4
CSL	Component Sub-Layer
CTD	Cell Transfer Delay
CUG	Closed User Group
CUSF	Call Unrelated Service Function
DBR	Deterministic Bit Rate
DDI	Direct Dialing In
DHA	Dialogue Handling
DPC	Destination Point Code
DSMCC	
DSS1, 2	Digital Subscriber Signalling system No. 1, 2
ECR	Explicit Cell Rate
FISU	Fill-In Signal Unit
GCRA	Generic Cell Rate Algorithm
GT	Global Title

IAM	Initial Address Message
IDN	Integrated Digital Network
IMT-2000	International Mobile Telecommunications – year 2000
IN	Inteligent Network
INAP	Intelligent Network Application Part
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
LI	Length Indicator
LSSU	Link Status Signal Unit
MBS	Maximum Burst Size
MCR	Minimum Cell rate
MSN	Multiple Subscriber Number
MSU	Message Signal Unit
MTP	Message Transfer Part
NI	No-increase Indication
N-ISDN	Narrowband-Integrated Services Digital Network
N-ISUP	Narrowband-ISDN User Part
NNI	Network Network Interface
NPC	Network Parameter Control
NSP	Network Service Part
OPC	Originating Point Code
OSI	Open Systems Interconnection
PC	Point Code
PCM	Pulse Code Modulation
PCR	Peak Cell Rate
PDU	Protocol Data Unit
PH	Physical Header
POTS	Plain Old Telephone Service
PNNI	Private Network Network Interface
PSPDN	Public Switched Packet Data Network
PVC	Permanent Virtual Connection (Channel)
PVPC	Permanent Virtual Path Connection
ROS	Remote Operations Service
ROSE	Remote Operations Service Element
RPC	Remote Procedure Call
SAAL	Signalling Asynchronous Adaptation Layer
SBR	Statistical Bit Rate
SCCP	Signalling Connection Control Part
SCEF	Service Creation Environment Function
SCF	Service Control Function

SCR	Sustainable Cell Rate
SDF	Service Data Function
SLS	Signalling Link Selection
SMAF	Service Management Access Function
SMF	Service Management Function
SRF	Specialised Resource Function
SS7	Signalling System No. 7
SSCF	Service Specific Coordination Function
SSCOP	Service Specific Connection Oriented Protocol
SSF	Service switching Function
SSN	Sub-System Number
STP	Signalling Transfer Point
SU	Signal Unit
SUB	Sub-addressing
SVC	Switched Virtual Channel
TC	Transaction Capabilities
TCAP	Transaction Capability Application Part
TCP/IP	Transport Control Protocol/Internet protocol
TFA	Transfer Allowed
TFP	Transfer Prohibited
TSL	Transaction Sub-Layer
UBR	Unavailable Bit Rate
UNI	User Network Interface
UPC	Usage Parameter Control
UPT	Universal Personal Telecommunications
UUS	User-to-User Signalling
VBR	Variable Bit Rate
VC	Virtual Channel
VCI	Virtual Channel Identification
VPI	Virtual Path Identifier
VPN	Virtual Private Network

ANNEX A

Other ATM Signalling Protocols

The ATM Forum has developed its own suite of NNI signalling protocols:

- Private Network Interface v1.0 (PNNIv1.0).

- Broadband Inter Carrier Interface (B-ICI) which is an ATM Forum endorsement of some B-ISUP Q.27xx recommendations. While B-ISUP is viewed by the ITU-T as an intra network interface as well as an inter network interface, B-ICI is considered as an inter network interface only (i.e. B-ICI should only be used at administrative boundaries).

- ATM Inter Network Interface (AINI) which was recently approved and provides means of interworking between PNNI and B-ISUP.

Actually the PNNI specification defines two different protocols:

- **PNNI Routing** which is a routing protocol for an ATM network; similar features also exit in IP networks. *PNNI Routing* is a link state routing protocol. It gives state and topology information to PNNI nodes and enables them to dynamically route ATM VCs and VPs. *PNNI Routing* is extremely scalable and was specified to work just as well in small as in very large networks.

- **PNNI Signalling** is the actual signalling protocol. *PNNI Signalling* is almost the same as UNI 4.0 signalling. Since UNI 4.0 is based on DSS2 recommendations, one can consider that so is *PNNI Signalling*. There is no UNI 4.0/PNNI interworking specification since as a result of UNI 4.0 and *PNNI Signalling* close relationship, interworking between the two is quite straightforward.

PNNI is not an SS7 network signalling protocol. It is transported on ATM VCs using the Signalling AAL (S-AAL). There is no support of ISDN services in PNNI, and no ISUP/PNNI interworking specification. With the existence of *PNNI Routing*, this is the main difference between PNNI and B-ISUP.

PNNI supports Switched VCCs, Switched VPCs, and Soft PVCs. Regarding Traffic Management, PNNI does not support ITU-T Recommendations like I.371 and I.356 as is. Instead, it is based on the ATM Forum Traffic Management v4.0 (TM 4.0) specification and thus supports Constant Bit Rate (CBR), real-time Variable Bit Rate (rt-VBR), non real time Variable Bit Rate (nrt-VBR), Available Bit Rate (ABR) and Undefined Bit Rate (UBR).

Finally, PNNI does not support native E.164 numbers and can only route calls on ATM End System Addresses (AESA).

The PNNI, B-ICI and TM 4.0 specifications can be found on the ATM Forum web site:
http://www.atmforum.com.

ANNEX B

DSS2 and the ATM Forum UNI signalling specifications

B.1 Introduction

The ATM Forum is working in parallel with ITU-T on the specification of UNI ATM signalling. This has led to the publication of UNI 3.1 (September 1994) and UNI 4.0 (July 1996). The scope of this Annex is to give a general overview of the differences at the functional level between the DSS2 and the ATM Forum UNI signalling systems. In particular, the intent of this Annex is not to provide the detailed differences between the protocols themselves and to identify possible inter-operability issues, but rather to give the reader the list of signalling functions provided in UNI 3.1 and UNI 4.0.

B.2 UNI 3.1 signalling specification

The following functions are provided in UNI 3.1:

- Point-to-point SVCs (cf DSS2 Q.2931 with some restrictions such as the non-support of the following DSS2 Q.2931 functions: Alerting phase, the Overlap receiving and Overlap sending procedures, the Notification procedure and the VP associated channel allocation).
- Point-to-multipoint SVCs (cf DSS2 Q.2971 with some restrictions such as the Alerting phase and the notification procedure).
- ATM traffic capabilities such as Constant Bit Rate (CBR) service (cf DSS2 Q.2931), Variable Bit Rate (VBR) service (cf DSS2 Q.2961.1), Unspecified Bit Rate (UBR) service (this service is not specified in DSS2), tagging (cf DSS2 Q.2961.1).
- QoS class indication (ITU-T Q.2931/Q.2965.1 QoS class 0 with ATM Forum specific QoS classes 1 to 4).

B.3 UNI 4.0 signalling

The UNI 4.0 signalling specification (see af-sig-0061.000 and its addendum af-sig-0076.000) provide the following signalling functions:

- Point-to-point SVCs. This function is described as a delta specification to DSS2 Q.2931.
- Point-to-multipoint SVCs. This function is described as a delta specification to DSS2 Q.2971.
- Generic identifier transport (DSS2 Q.2941.1). This function is described in the delta specification to Q.2931 (point-to-point SVCs) and in the delta specification to Q.2971 (point-to-multipoint SVCs).
- ATM traffic capabilities such as Constant Bit Rate (CBR)/Deterministic Bit Rate service (cf DSS2 Q.2931 and Q.2961.2), Variable Bit Rate (VBR)/Statistical Bit Rate service (cf DSS2 Q.2961.1 and Q.2961.2), Unspecified Bit Rate (UBR) service (this service is not specified in DSS2), tagging (cf DSS2 Q.2961.1), Available Bit Rate service (cf DSS2 Q.2961.3 with minor extensions) and Frame discard (not specified in DSS2).
- QoS class indication (cf DSS2 Q.2931/Q.2965.1 QoS class 0 with ATM Forum specific QoS classes 1 to 4).
- Switched Virtual Paths (cf DSS2 Q.2934). This function is described in the delta specification to Q.2931.
- Signalling of individual QoS parameters (CTD, CDV, CLR) (cf ITU-T Q.2965.2 which is under definition).

- Negotiation of connection characteristics (cf DSS2 Q.2962).
- Supplementary services (based on DSS2 DDI, MSN, SUB, CLIP, CLIR, COLP, COLR and UUS).
- Leaf initiated Join (LIJ) (not defined in DSS2).
- ATM any-cast (not defined in DSS2).

The following ITU-T DSS2 signalling functions are not specified in UNI 4.0:

– Signalling for the ABT service (i.e. DSS2 Q.2961.4), ITU-T SBR2/SBR3 capabilities (cf DSS2 Q.2961.6);

– Signalling of CDVT values (i.e. DSS2 Q.2961.5);

– Modification of connection characteristics (i.e. DSS2 Q.2963.1, Q.2963.2 and Q.2963.3);

– Explicit signalling of I.356 QoS classes (i.e. DSS2 Q.2965.1 and Amendment 3 of Q.2931);

– B-ISDN frame relay bearer service (i.e. DSS2 Q.2933);

– Call priority (i.e. DSS2 Q.2959) and Closed User Group (i.e. DSS2 Q.2955.1);

– Generic functional protocol (i.e. DSS2 Q.2932.1);

– Look-ahead (i.e. DSS2 Q.2964.1);

– Signalling of AAL type 2 and the associated parameters (i.e. DSS2 Q.2931 Amendment 2).

CHAPTER 5

5 Synchronization techniques and methods

5.1 Introduction

Synchronization plays a critical role in modern communications, and the special timing aspects of communication engineering are an important part of the telecommunication engineers' training. Since PCM was invented and introduced into global communication networks, synchronization has been one of the most essential factors in the communication system design and maintenance.

In a digital transmission system, synchronization is an essential receiver function. Here binary information is converted by means of a modulator into a continuous-time signal, which is sent over the transmission channel. A digital receiver extracts the information sequence from a discrete-time signal obtained after sampling and quantizing the distorted signal presented to the demodulator. At the receiver, accurate timing recovery is critical to obtain performance close to that of the optimal receiver [1]. Nowadays, synchronization has evolved into an autonomous research area that has been studied in depth over recent decades.

The goal of communication network synchronization is to provide two necessary conditions for real time digital information exchange between users:

– The *continuity*, which means that bit rates must be the same for interconnected terminals;

– The *integrity*, which means that information elements (bits, bytes or blocks) must been received in the sink in the same order as they were sent from the source.

In other words, *synchronism* signifies a common time reference for transmitter and receiver. Correspondingly, synchronization is a crucial operation in the reception of data signals. Two categories are commonly recognized [2,3]:

– Signal synchronization, where symbol timing, carrier phase, and carrier frequency of a receiver are aligned with an incoming signal,

– Frame synchronization, where burst, packet, word, code, or frame boundaries are recovered from a received data stream.

Correspondingly, in practical applications synchronization tasks for communications are divided into three levels, such as [4]:

– The carrier synchronization;

– The bit timing;

– The frame (or block) alignment.

The first task deals with the carrier frequency (phase) tracking while in operation. The second task is important for modern digital communications and dealing with digital (bit) sequences. It is known as "*Network Synchronization*" or "*Synchronization Network*", "*bit timing*" or "*bit synchronization*". This is the main subject of this chapter. Finally, frame (block) alignment is the synchronization task for packetised transmission systems.

Two types of synchronizer structures can be distinguished with this.

- DA synchronizers use the receiver's decisions (decision-directed) or a training sequence in computing the timing estimates.
- NDA structures operate independent of the transmitted information sequence.

5.2 Synchronization of the transmission systems

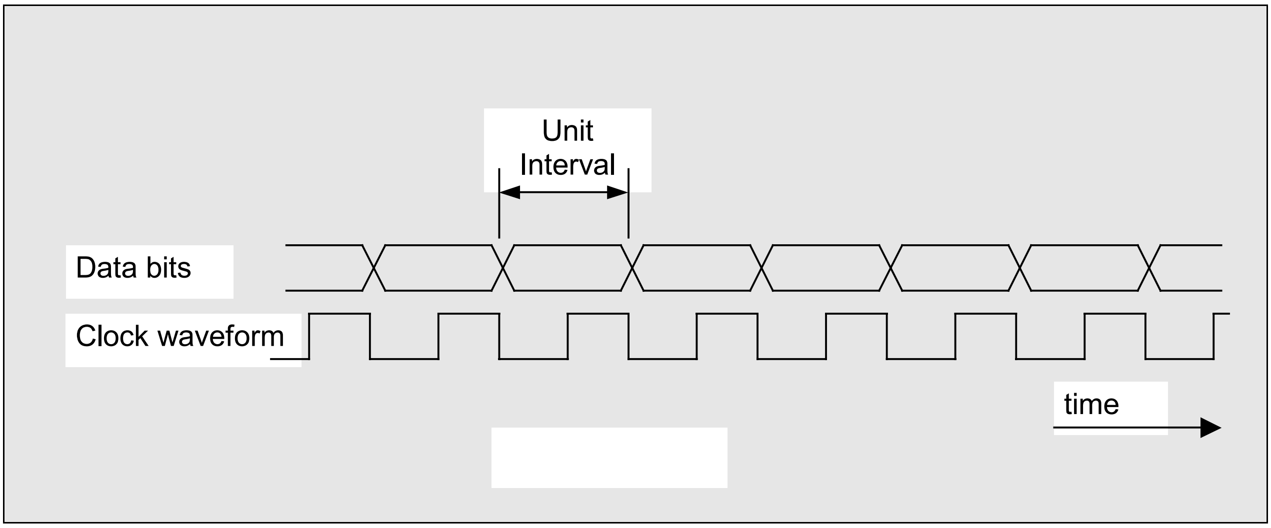
At the early stage of PCM development, local synchronization of the point-to-point communication was the problem. As usually, digital signal consists of two elements, such as:

– a stream of binary digits (bits); that is data;

– the instants in time (discrete points of time) at which the bit occur.

In practice, the second element is a timing signal known as *a clock*. Figure 5.1 shows example of the data signal with associated clock. As may be seen, unit intervals are uniformly distributed in time so that each unit interval contains only one bit. The negative-going edge of the clock pulse is a boundary between bits in a continuous binary stream.

Figure 5.1 – Example of data signal and associated clock



5.2.1 The Clock Signal Extraction from Binary Data stream

Timing (clock) is necessary for data processing in both the transmitter and in the sampling of data bits in the receiver. There is no need to transmit the *clock waveform* jointly with the data as the latter contains the *clock signal* information [5]. Figure 5.2 illustrates this finding in the time domain. Considering this Figure in detail; the modulo two sum $\{D(t) \oplus D(t + t_D)\}$ of the data signal $D(t)$ and delayed signal $D(t + t_D)$ gives a pulse sequence spaced with random intervals due to random nature of the signal edge time points. In Figure 5.2, as an example the delay time t_D is taken as 1/4 part of a unit interval T. Signal $d(t) = \{D(t) \oplus D(t + t_D)\}$ may be presented as algebraic sum of deterministic and random parts $s(t)$ and $r(t)$

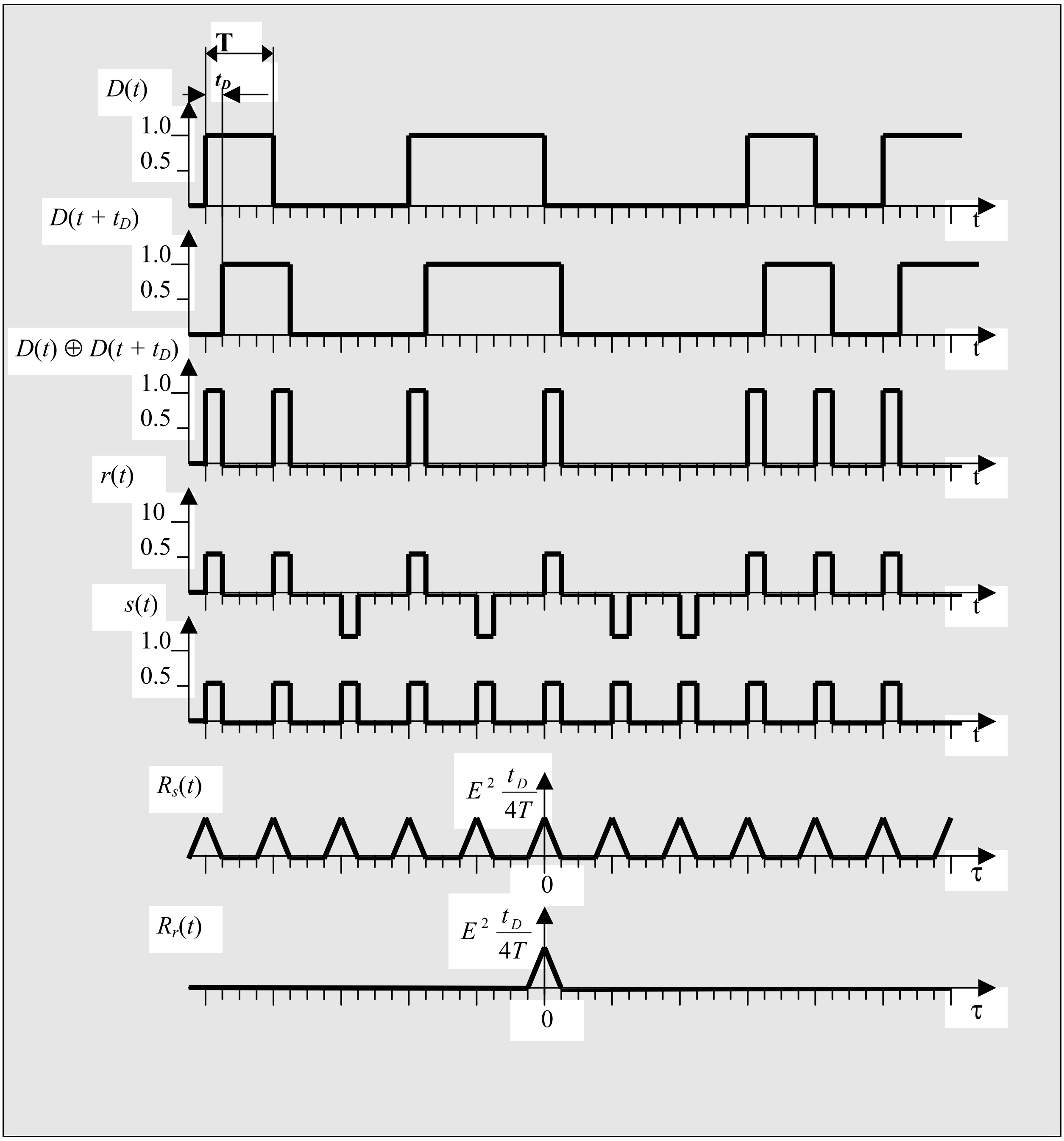
$$d(t) = s(t) + r(t)$$

Note, the deterministic (systematic) part $s(t)$ is the desired periodic clock signal sequence as pulses are separated by an equal time unit distance.

As can be seen from Figure 5.2, both signals $s(t)$ and $r(t)$ are sequences of rectangular pulses. It is known that covariance of the periodic pulse sequence $s(t)$ is also a periodic function R_S (τ) with the same repetition period T. However the covariance of the random signal $r(t)$ is the non-periodic function $R_r(\tau)$ which has a triangular shape in the time duration $-t_D \le \tau \le t_D$. Total covariance of the signal $d(t)$ is presented as superposition of particular covariances

$$R_D(\tau) = R_s(\tau) + R_r(\tau)$$

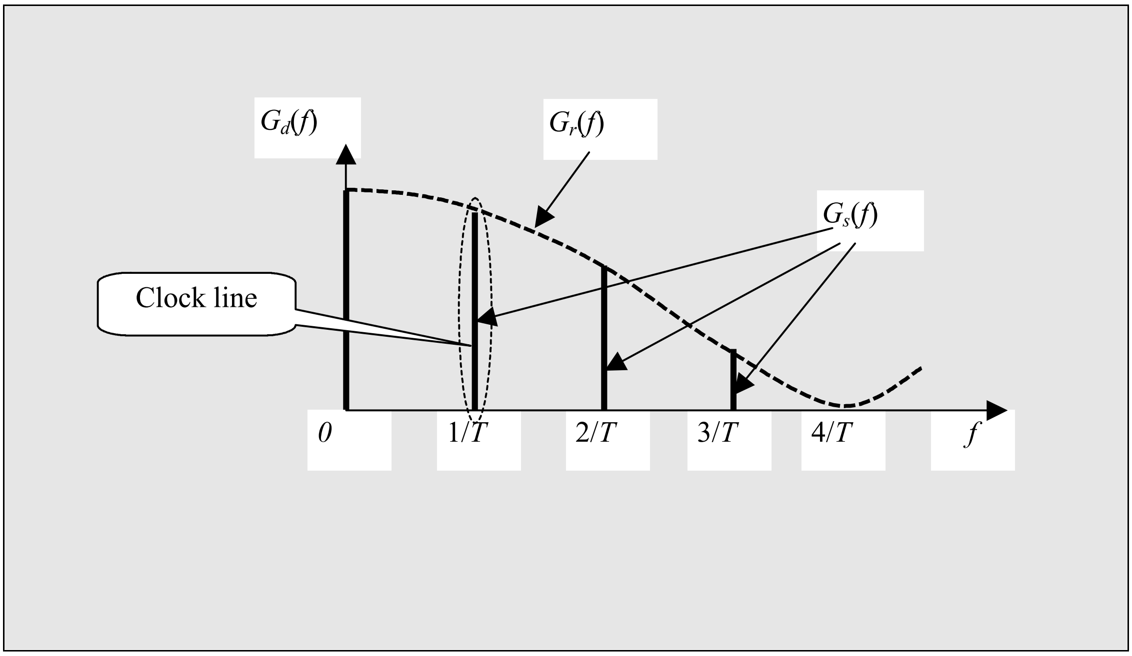
Figure 5.2 – Extraction of timing information from data



Power spectrum density $G_d(f)$ of the signal $d(t)$ is defined through covariance $R_D(\tau)$ by Fourier transform (Wiener-Khintchin transform) based on which we realize that $G_d(f) = G_s(f) + G_r(f)$, where the power spectral density $G_s(f)$ of the signal $s(t)$ has discrete shape due to its periodic nature, and that $G_r(f)$ of noise $r(t)$ is a continuous function due to its random origin. Figure 5.3 shows, for example, total $G_D(f)$ and particular spectrums $G_s(f)$ and $G_r(f)$ of the signal $d(t)$. Here function $G_r(f)$ is considered as envelope of the total spectrum.

Note the key properties of the spectrum (Figure 5.3). The spectral line with $n = 1$ represents the power of the sine signal with the frequency $1/T$ and is considered as the clock signal. By extracting this sine signal from the total $d(t)$ one may form a correspondent clock signal intended for synchronization.

Figure 5.3 – Power spectral density



Thus, there is no need to send both data and associated clock signals through transmission systems because the timing signal may be recovered from the data stream by means of $\{D(t) \oplus D(t + t_D)\}$sequence generation and the clock frequency component $f_0 = 1/T$ filtering from the spectrum of this pulse sequence. Two opportunities are realized for this case based on:

– Narrow bandpass (BP) filter (may be an optimal filter) with central frequency $f_0 = 1/T$,

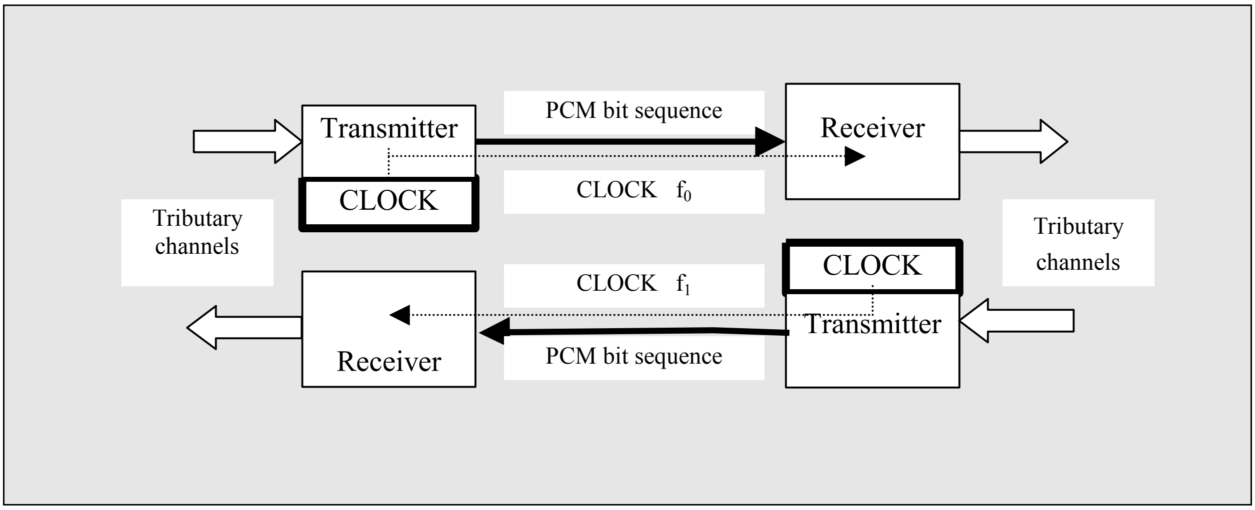
– Phase lock loop (PLL) with low pass (LP) or optimal filter.

5.2.2 Point-to-point synchronization

Usually, a bi-directional digital line provides transmission for digital communications based on two digital transmission channels which are the means for unidirectional digital transmission of digital signals between two points. For this case, the essence of point-to-point synchronization is illustrated by Figure 5.4 [6].

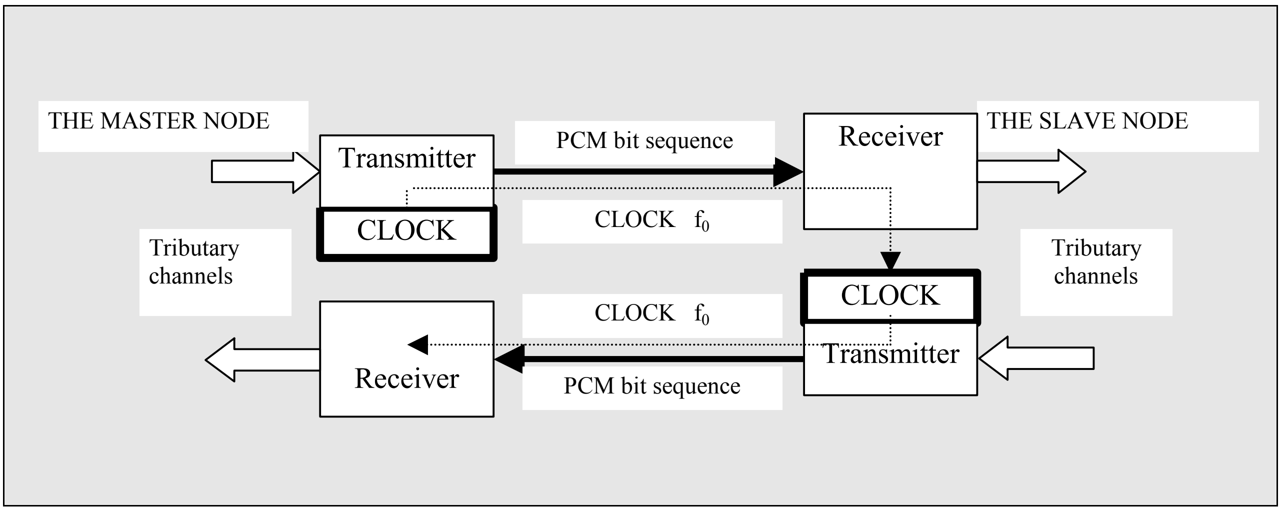
The simplest synchronization technique usually applied is that used by the Plain Old Telephone Service (POTS) in the early days of PCM transmission. Here independent synchronization of each channel with clock rates f_0 and f_1 is used in accordance with the following structure (Figure 5.4a).

Figure 5.4 a) – Independent synchronization



It is known that the performance of communications systems are dramatically improved with implementation of digital methods. At the same time, once digital communication networks are implemented, then master and slave synchronization network nodes require accurate timing of the data. In this case (Figure 5.4b), the transmitter of the main node sends a PCM data stream with a clock rate f_0 to the slave node.

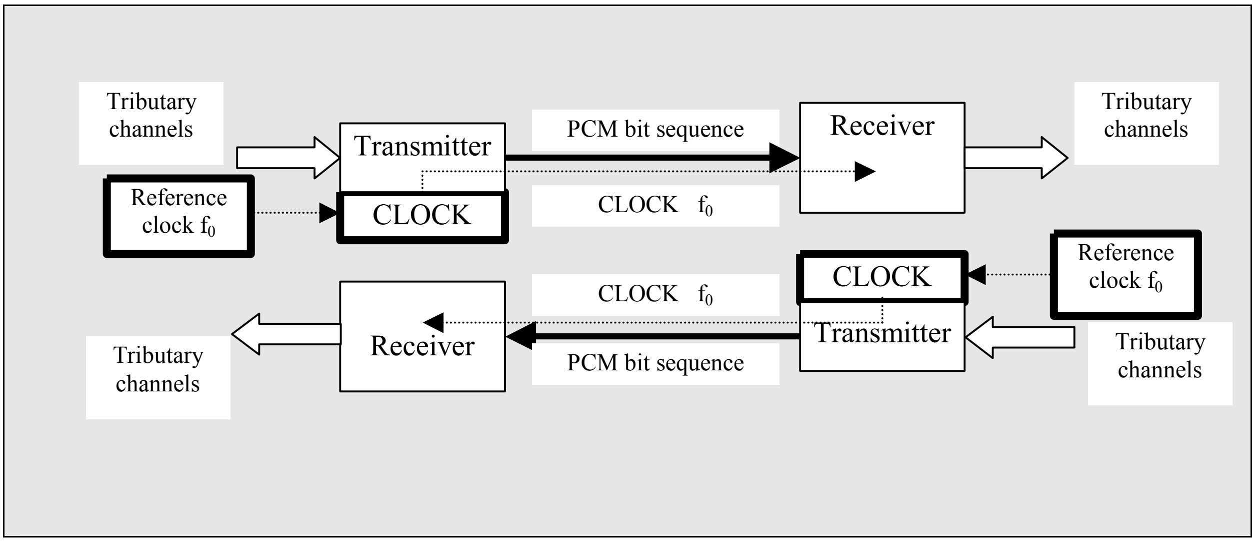
Figure 5.4 b) – Slaved synchronization



Here the receiver at the slave node derives the clock signal from the data by using the methods described above. The clock signal extracted is used as a reference for adjusting the clock of a slave node transmitter. Thus, both PCM data streams are transmitted with the same clock rate f_0.

Finally, Figure 5.4c shows a typical structure of a synchronous network or networks synchronized by an external accurate master clock. Here PCM bit sequences are transmitted through the network with a clock rate f_0 in both channels (in both directions). With respect to external master clock, both clocks of the first and second transmitters are slaved. In this case, the *reference timing signal* must be distributed amongst all the telecommunication network nodes that are included in this *synchronization network* planning. Below we will consider this question in detail.

Figure 5.4. c) – Synchronous network



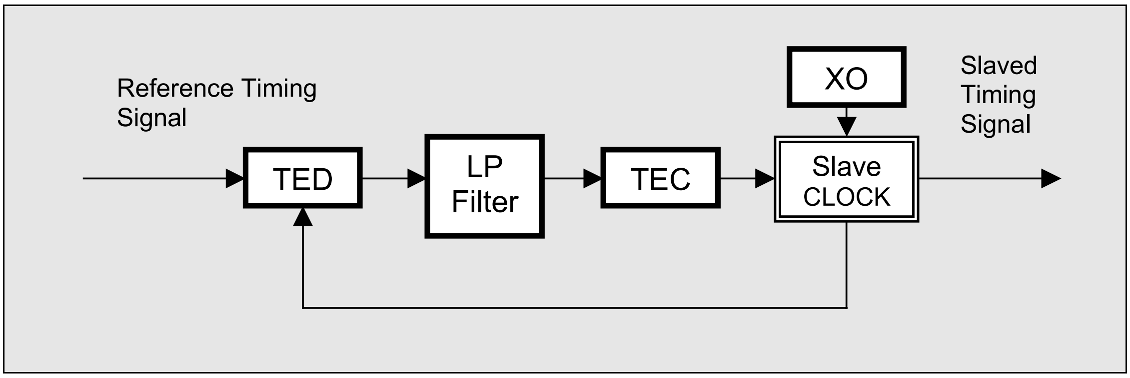
5.3 PLL based techniques

In non-synchronized communication systems, a free-running oscillator determines the sample instants at the receiver, and digital processing of the sample sequence provides time correction. In synchronized systems, in order to realize data synchronization in slave nodes it is necessary to get correspondent circuits embedded into transmitting equipment. Such circuits are of feedback type and based on a digital PLL design intended for phase (time) disciplining of a slave clock via an external reference signal from a master clock. Generally, there are two recognized PLL circuits intended for 1) discrete time correction, 2) continuous time correction.

5.3.1 Slave clock correction in discrete time

Figure 5.5 shows the generalized structure of a PLL feedback circuit intended for discrete time correction of the slave clock using a discrete time reference signal of the master clock.

Figure 5.5 – PLL feedback circuit with discrete time correction

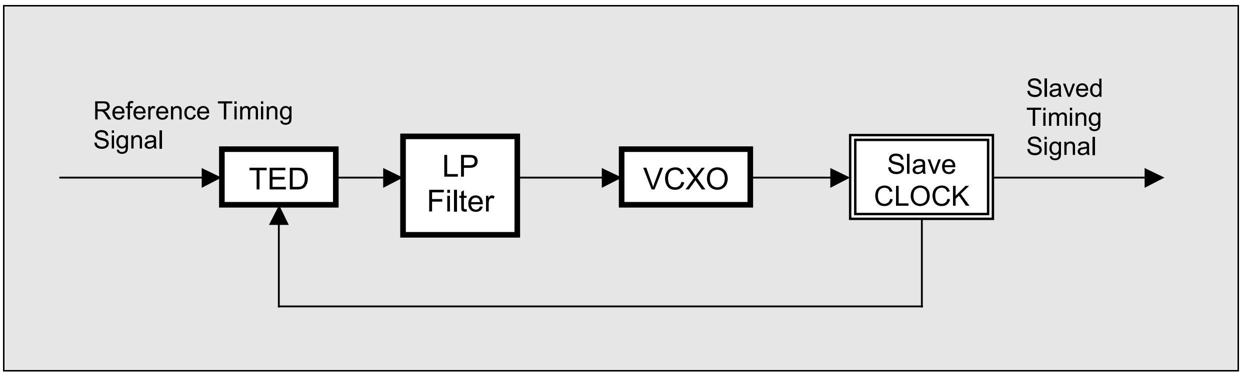


Essentially, the circuit consists of a Timing Error Detector (TED) and Timing Error Corrector (TEC). Initially, the slave clock provides a timing signal for the transmitter based on the external crystal oscillator (XO) signal. Naturally, the signal formed does not coincide with that of the reference source because of frequency drift and phase shift. To eliminate this timing error, the TED compares the time signals of the reference source and the slave clock, and forms an error signal corresponding to the time difference between the time sources. The error signal passes through the LP (Optimal) filter to the TEC, which makes the correction to the slave clock timing . Because of the relatively low cost of the XO and its low noise jitter, discrete time correction is widely used now for many practical applications.

5.3.2 Oscillator disciplining in continuous time

In the continuous-time domain, timing adjustment can be performed by controlling the sampling phase of oscillator. Figure 5.6 displays a correspondent structure of a PLL based feedback circuit intended for direct tracking of the slave oscillator in continuous time via a reference signal from the master clock.

Figure 5.6 – PLL feedback circuit with reference signal control



The circuit operates in a similar way to that considered in Figure 5.5. Here the continuous signal of VCXO is compared with the reference one and a filtered error version of the TED output is fed to the device that determines the sampling instants (e.g., a VCXO). In this way, the error signal passes through a LP filter and directly controls the oscillator frequency to eliminate the phase (time) shift between the two signals. Because of its relatively high cost and noise jitter, the use of a VCXO as a discrete component is to be avoided, if possible.

5.4 Timing correction

It follows from the Figure 5.4 analysis that accurate timing recovery at the receiver is critical to obtain performance close to that of the optimal receiver. Receiver synchronization is based on statistical methods of timing signal estimation. A major part of synchronizer algorithms can be related to the Maximum Likelihood (ML) criterion [3] according to which the estimates of timing parameters maximize the likelihood function. Correspondingly, the likelihood function depends on the signal modulation format and the statistical properties of the noise added to the signal. The timing estimates maximize the likelihood function, which is obtained by averaging over the random information variables.

Timing estimation may be done for both multitone modulation and single carrier signals.

According to the ADSL standard [7], a single unmodulated carrier must be received for timing extraction. The frequency of this carrier (called the pilot) is fixed. In comparison with the multitone signal, synchronization by means of a pilot has several disadvantages [1].

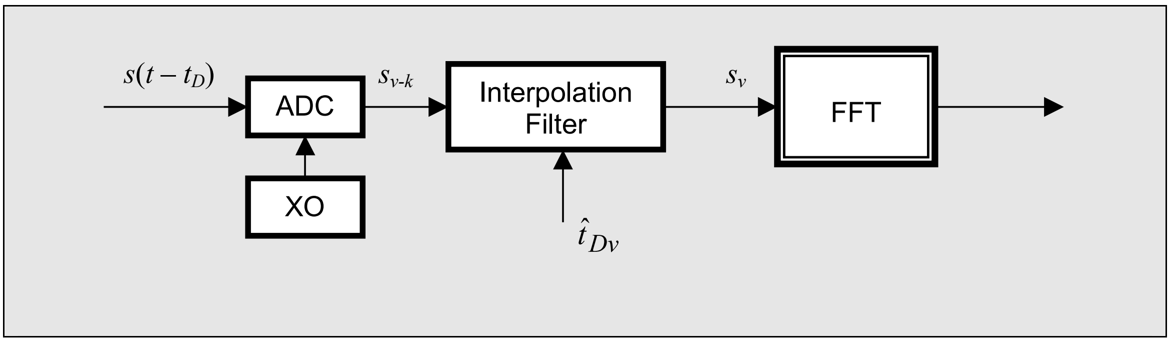
- Because the timing estimator exploits only a fraction of the received signal power, the variance of the timing error estimate is larger than the variance obtained when taking the multicarrier approach.
- To achieve the same error variance, the closed loop bandwidth of the feedback PLL synchronizer must be decreased. As a result, the capability of the synchronizer to track frequency offset variations is affected.
- When using an unmodulated tone, the timing error estimate becomes sensitive to a systematic, non-time-varying disturbance such as temperature, aging, etc.
- If the SNR at the pilot frequency happens to be low, the estimator produces unreliable timing estimates or much more time is necessary for timing signal estimation.

We consider below major approaches for timing correction in time and frequency domains based on timing error estimates.

5.4.1 Error correction in discrete time domain

For many years, many methods have been developed to realize a fractional delay in the discrete time domain [8]. Usually, the timing correction is performed by means of a finite impulse response interpolation filter (Figure 5.7).

Figure 5.7 – Timing correction in discrete time domain

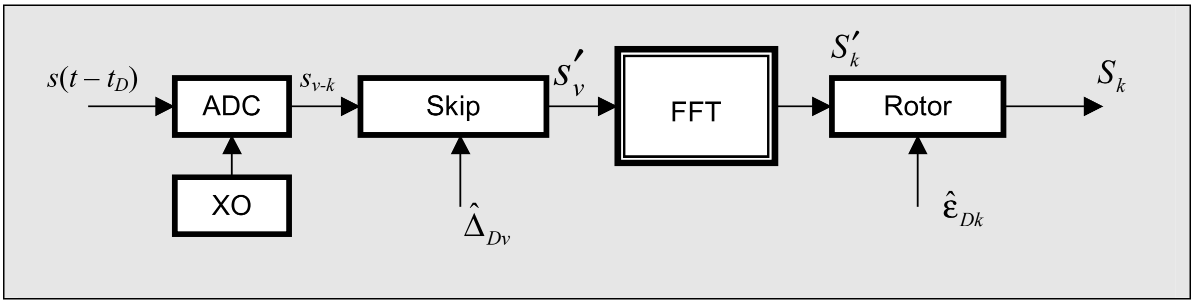


The coefficients of the filter depend on the time error t_D to be corrected. The input signal $s(t - t_D)$ is transformed by the ADC into a digital form s_{v-k}, where the coefficient k corresponds to time delay t_{Dv}. Controlling the signal corresponding to the time error estimate $\hat{t}_{Dv}$ affects on the interpolation filter by changing its coefficients in a way which leads to the corrected signal appearing at the filter output. As a result, the signal spectrum at the output of the FFT receives the desired attenuation of the spectral amplitudes and rotation of the phases. For ADSL [7], this method turns out to be applicable. For very-high-rate systems, such as VDSL, this approach cannot be used because of the high sampling rate and computation complexity involved. In VDSL, however, one can use synchronizer circuits that exploit the delay-rotor property [1].

5.4.2 Error correction in frequency domain

In a case of a non-zero slave clock frequency offset, the time error t_D increases linearly in time. Whereas ideally the timing error correction should be different for each signal sample, the TEC will correct the timing for each sample over the same delay. Because of the delay-rotor property, this timing correction can be performed in the frequency domain by rotating the FFT outputs as shown in Figure 5.8 [1].

Figure 5.8 – Timing correction in frequency domain

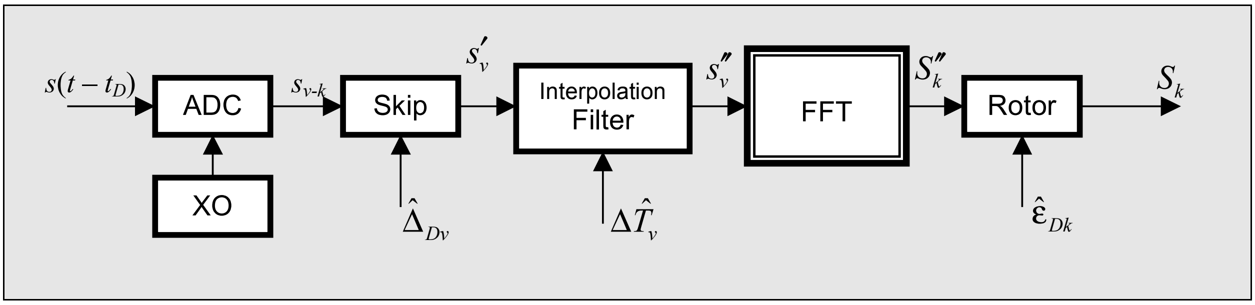


The skipping corrector makes discrete signal control on the integer part $\hat{\Delta}_{Dv}$ of the total estimate of time error $\hat{t}_{Dv}$ so that, in the simplest case, the error to be corrected is ± 1. The fractional part $\hat{\varepsilon}_{Dk}$ of $\hat{t}_{Dv}$ is corrected by rotation of the spectral components S'_k of the signal which leads to the desired spectral shape S_k at the output of the rotor. A merit of this method is its low complexity. No oversampling at the receiver input is required, and, for each carrier, correction is performed by a single complex multiplication. This method works well only when a small frequency offset can be guaranteed. This mandates very accurate (and hence more expensive) XOs at both transmitter and receiver sides. Alternatively, a low-cost, less accurate VCXO can be used to reduce the initial sample clock frequency offset. Correction of the residual offset is then performed in the digital domain [1].

5.4.3 Error correction in both time and frequency domains

Figure 5.9 depicts the time error correction based on both interpolation filter in time domain and spectrum rotation in frequency domain [1-3].

Figure 5.9 – Timing correction in discrete time and frequency domains



An interpolator filter is used to correct the error increment $\Delta\hat{T}_v$ in a similar manner to Figure 5.7 and skipping corrector realizes step timing error elimination as shown in Figure 5.8. Finally, the rotor provides spectrum phase rotation to obtain the desired accuracy of the timing error in accordance with Figure 5.8. The main advantage of this method is that the interpolator has to correct only for small timing errors. For example, for the frequency offset of 100 ppm and code $N = 256$, the interpolator has to correct a maximal timing error of 2.56×10^{-2} samples. A disadvantage of the presented technique is that it exploits the delay-rotor property, which is only valid under specific conditions.

5.5 Network synchronization

5.5.1 Synchronization of telephone exchanges in the PSTN

Initially PCM digital transmission systems were introduced with their own synchronization subsystems intended for their equipment internal needs (Figure 5.4a). Figure 5.4b pertains to the situation where the digital exchange was appearing in the master network node. As digital centres were deployed, the problem of synchronous working appeared for effective-cost interaction. According to a pragmatic approach, the same clock frequency is required in the output and input tributary channels in the master node. So, the internal clock of the slave node PCM equipment must be locked to the internal clock frequency f_0 of the master node PCM equipment. Moreover, both clocks are to be used as synchronizing signal sources for the cooperating digital equipment in corresponding nodes. As shown in Figure 5.4c, since the digital exchange is introduced into all nodes of existing telephone networks, then a reference clock for all equipment in each switching office is used so that PCM equipment internal clocks are locked to this reference clock also. With this comes the problem with a reference clock frequency being delivered to geographically spaced switching nodes. Synchronization network planning is solving this and we may view the AT&T Switched Network Synchronization system realized during 1980s as the classical example in this regard.

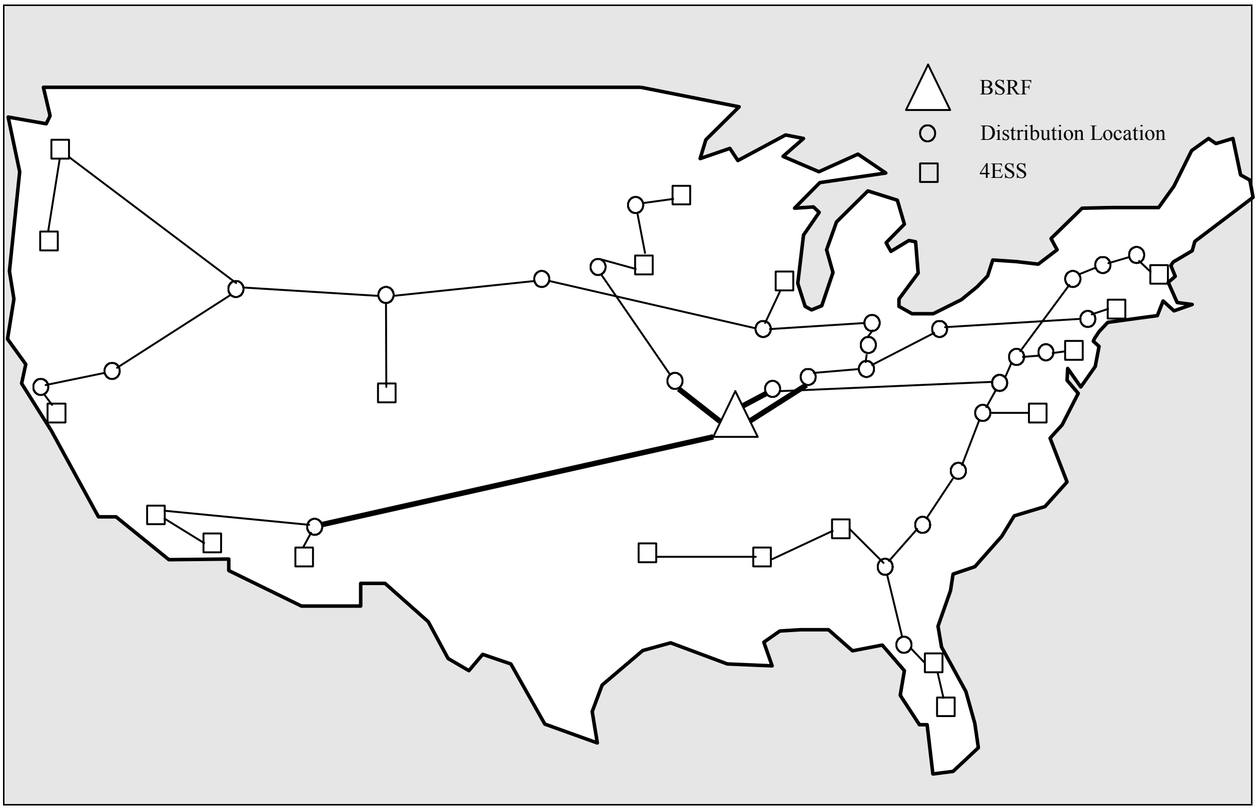
The heart of AT&T′s synchronization network is a caesium clock ensemble known as BSRF located in the country “heaviness centre”. The reference frequency from BSRF is distributed via analogue radio and coaxial systems to the 4ESS digital switching offices as shown in Figure 5.10 [9]. Such a refined but costly distribution network requires additional expenses for skilled maintenance, so the needs for a synchronization network should be explained.

In a digital switching office, the clock system controls timing of the thousands of message trunks. Intermediate buffers are necessary for absorbing the time “jitter” and “wander” which are arising in transmission systems, and for the frame alignment necessary for error free time-division switching. The network contains a great number of interacting clocks and requires reliable frequency control of all clock sources within a limited range of permitted accuracy for acceptable service quality. Here intermediate buffer memories are being emptied/overflowed if the rate of incoming bit stream runs slowly/quickly with respect to that of the local clock. As a result, stored data is being either read twice or lost. This is the reason why such a bit repetition or delay is called a “slip”. To avoid this effect, buffer control is implemented for 8-bit word slips only in E1 digital signals so that a mean rate of the slips is considered as an important performance characteristic of the network quality. It is obvious that this mean is tended to be as low as possible based on the following effects:

– the introduction of the more rigid tolerance for the network clock frequency accuracy leads to relatively small buffers memory size,

– reducing requirements for the network clock frequency accuracy leads to a rise in buffers memory size.

Figure 5.10 – AT&T Reference clock distribution



Buffer memory size rises result in a proportional increase in the transmission delay through the network, and this important network quality performance factor should be limited. Delay accumulation in various services leads to performance violations, in telephony for instance, a delay of more than 100ms causes problems with echo suppression. Paper [10] addresses the necessary recommendations for works provision with respect to the timing accuracy improvement of the network clocks (Table 5.1).

Table 5.1 – Synchronization objectives and references

The subject be synchronized	Objectives	references
Internal PCM equipment clock (absolute accuracy)	50×10^{-6}	ITU-T G.703 Recommendation
POTS most critical equipment (absolute accuracy)	$\sim 1 \times 10^{-9}$	**AT&T's data [9]**
Network international interface (accuracy with respect to UTC)	1×10^{-11}	ITU-T G.811 Recommendation
BSRF for the transport network (accuracy with respect to UTC)	1×10^{-12}	**AT&T's data [9]**
The transport network Primary node (stability)	7×10^{-13}/day 1×10^{-13}/week	**Bellcore objectives [9]**
The transport network Secondary node (stability)	8×10^{-13}/day 3×10^{-13}/week	**Bellcore objectives [9]**

5.5.2 Dynamics of the ITU-T activity in synchronization aspects of communications

In the past, timing aspects of the PSTN have been regarded as a subject for specialists in the field, and have not been normally considered as a part of general telecommunications training. After the introduction of SDH, the basic starting points of time and frequency have been re-considered and reduced into the formulas recommended to avoid problems for network operators.

Extensive network growth (from 1993 to 2000 the number of Central Offices has doubled) and the creation of new correspondent technologies (SDH/SONET, ATM etc.) based on the use of enhanced services (credit cards, 800 services etc.) requires a trouble-free deployment of the new network technologies. Synchronization is a necessary condition for network operation and management with high service quality at low cost. The dynamics of international standardization body activities in the field of network synchronization reflects the network technology development. Note, the AT&T experience (Figure 5.10) was embodied into three CCITT Recommendations of Blue Book (1988):

G.810 "Timing and synchronization considerations";

G.811 "Timing requirement at the outputs of primary reference clocks suitable for plesiochronous operation of international digital links";

G.812 "Timing requirements at the outputs of slave clocks suitable for plesiochronous operation of international digital links".

Plesiochronous operation is a good decision for international interfaces. It helps to avoid the delicate problem of who should synchronize to whom at the level of a border gateway. With plesiochronous operation, we nominally get the same rate in any node but without adjustments. Figure 5.11 depicts a typical synchronization chain corresponding to the Recommendations and requirements for timing characteristics of node clocks (Table 5.2).

Table 5.2 – Timing characteristics of node clocks

CCITT Stratum Levels (1988)	Hold-in accuracy	Holdover accuracy		Normative reference
		Initial frequency offset	Frequency drift per day	
PRC	1.0×10^{-11}	–	–	G.811 (11/88)
TNC	–	5.0×10^{-10}	1.0×10^{-9}	G.812 (11/88)
LNC	–	1.0×10^{-8}	2.0×10^{-8}	G.812 (11/88)
CPE clock	5.0×10^{-5}	–	–	I.431 (11/88)

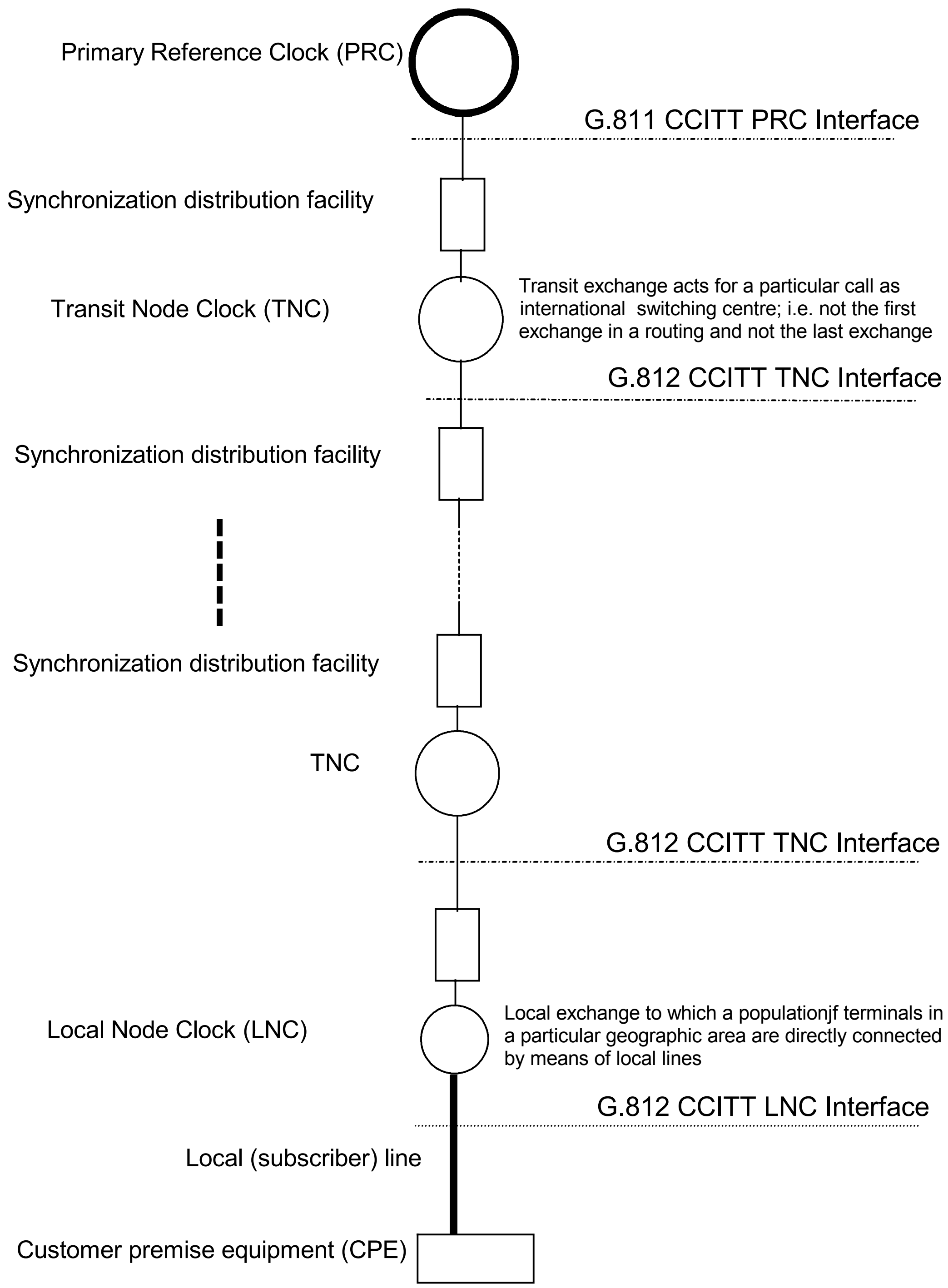
It should be noted that the ITU-T does not determine the accuracy of the TNC. Instead of accuracy, the performance for the holdover operation mode of the slave clock is determined. Holdover operation is started as soon as the reference signal is not available (lost). This means that a slave clock output no longer reflects an external reference. Generally, in the holdover mode of operation, the clock output control is based on stored data acquired while hold-in lock operation. Prediction of a slave clock frequency behaviour in time is most helpful here. It is based mainly on Kalman filter usage in PLLs and gives good results for hours or days. The modern modulation approach for crystal-based clocks [11] obtains good results for days, months, and years, and is now under development.

Correspondent Recommendations characterize the performances of node clocks at each level. In the particular case, the G.811 (11/88) Recommendation sets limits for long term PRC accuracy with respect to UTC at the 1×10^{-11} level which the USA's commercial caesium clocks exhibited in 1988.

Note – synchronization distribution facilities (Figure 5.11) have only been treated in isolated ways. Characterization of the links connecting node clocks requires a preliminary analysis of their proportions and correspondent efforts are now under consideration. Finally, to arrive at overall estimates of synchronization network performance, the standard definitions and measures for distribution links are given, which are easily combined with that for the clock.

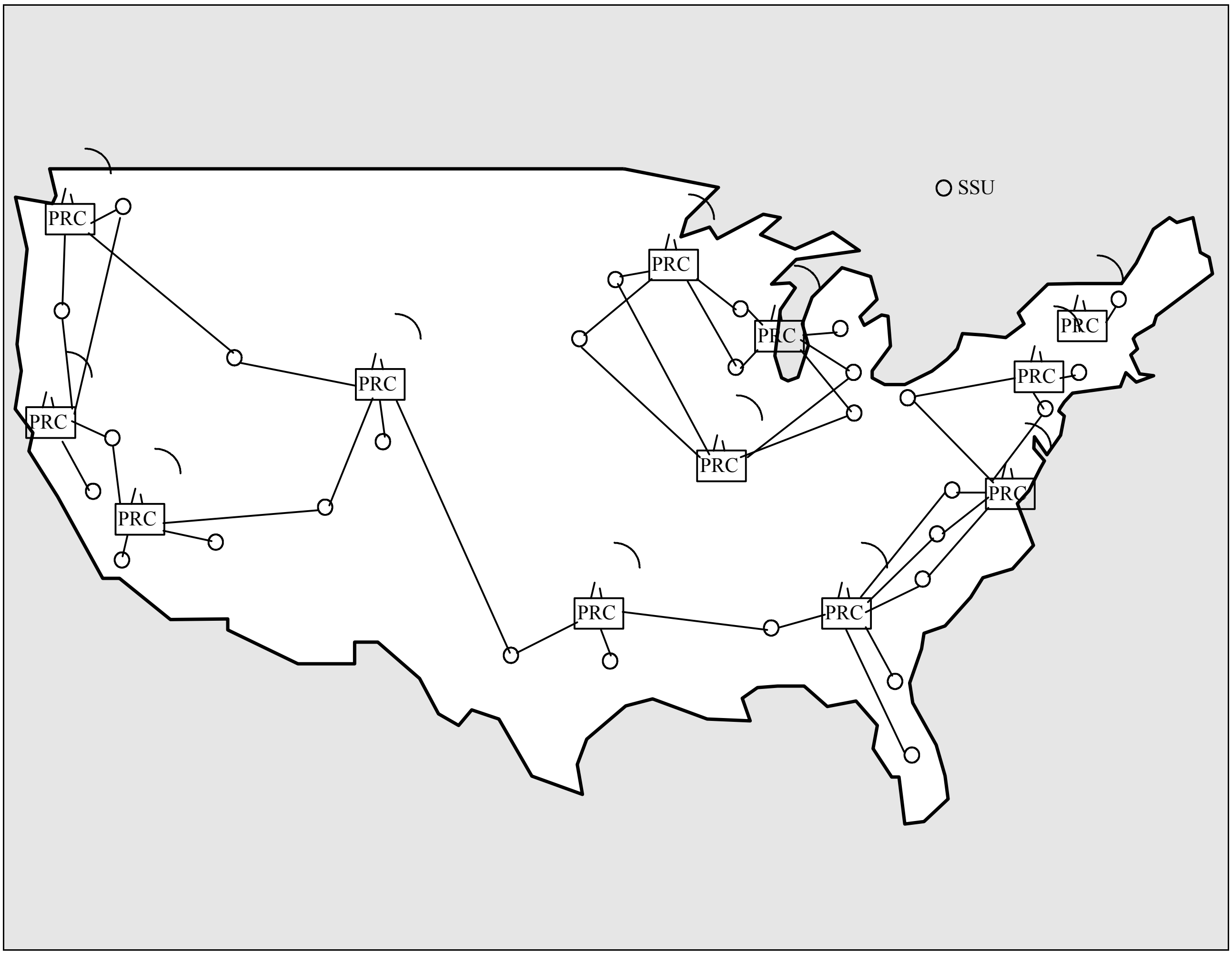
Synchronization of the PSTN ensures that all digital exchanges were ultimately slaved to the same master clock (see Figure 5.10). Nevertheless, in reality network operation is «stressed» when slave clocks receive a synchronizing signal from a PRC over a facility that has impairments. Accumulation of the facility-caused errors results in an inaccuracy increase of PRC frequency from 10^{-12} to 10^{-10} [9, 12]. Therefore, the clock distribution contributes a much larger portion of timing errors in a network than the clocks will, and such performance degradations are not acceptable for advanced transport network based on SONET/SDH technology.

Figure 5.11 – Typical synchronization chain



The new AT&T synchronization plan has been introduced during the 1990s improving accuracy performance by two orders of magnitude to avoid the long distance synchronization distribution deficiency. It is related to the "partly distributed PRC" synchronization network [13] (Figure 5.12).

Figure 5.12 – PRC synchronization



New clock distribution means are used for new planning. Namely, the best quality PRC signal is accessible worldwide due to GPS/GLONASS, a space-based radio positioning utility [14]. Instead of a single PRC, sixteen PRCs with Rubidium/GPS technology are distributed by an all-digital SONET transport network in which time-division transmission and switching functions are critical. Low cost, simple and reliable clocks to be equipped by the compact GPS receiver that satisfy the PRC requirements providing a frequency accuracy up to 3×10^{-12} (one day average) and a time accuracy of ± 150 ns with respect to GPS time [15]. The network of partly distributed PRC with CPS receivers to provide a long-term timing accuracy to nodes, is richly interconnected and has a good capability of verification at several levels to detect and resolve timing degradation before it impacts service.

This has been taken as the background for modern ITU-T Recommendations such as:

G.803 (06/97) "Architecture of transport networks based on the synchronous digital hierarchy (SDH)" – Section 8: "Architecture of synchronization networks";

G.810 (08/96) "Definitions and terminology for synchronization network";

G.811 (09/97) "Timing characteristics of primary reference clock";

G.812 (09/97) "Timing requirements of slave clocks suitable for use in node clocks in synchronization networks";

G.813 (08/96) "Timing characteristics of SDH equipment slave clocks (SEC)";

G.871 (02/99) "Synchronization Layer Functions".

Figure 5.13 – New Network synchronization

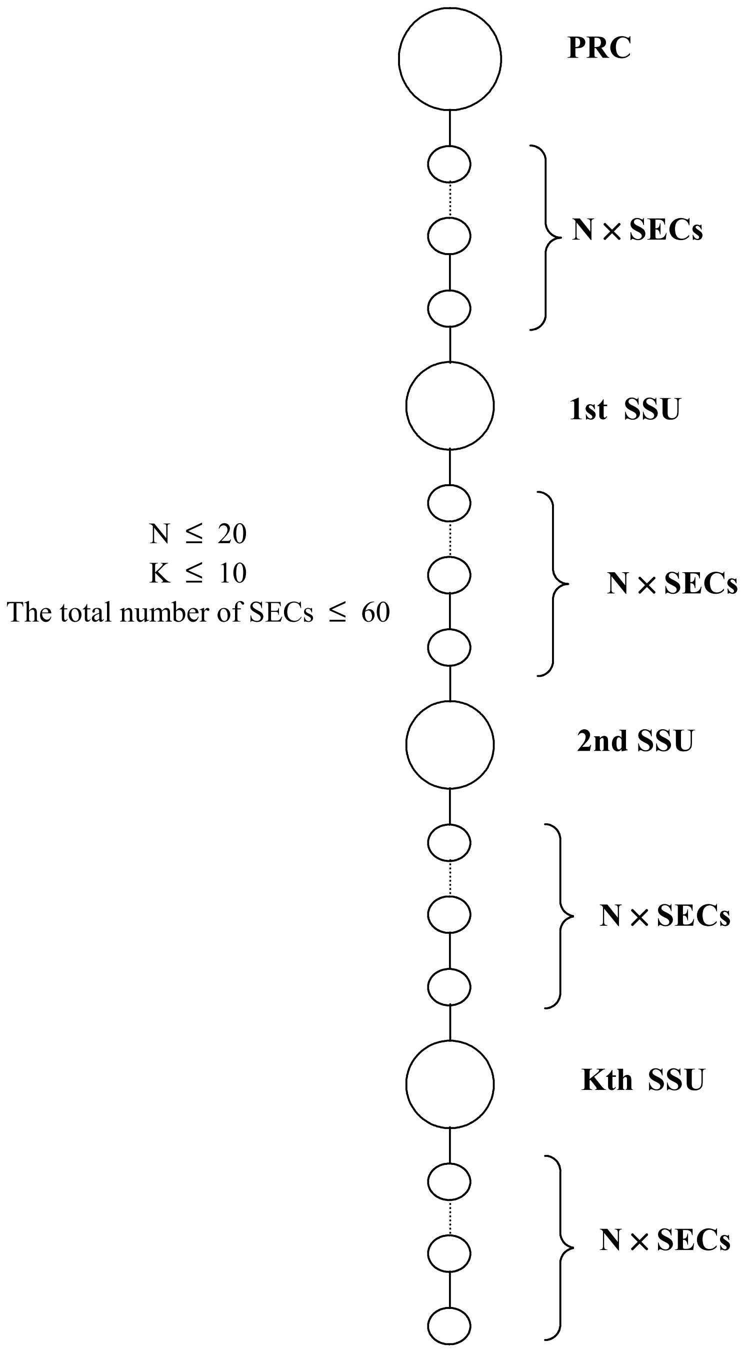


Figure 5.13 shows a chain that pertains to the new network synchronization facility, where node clocks are interconnected via *N* network elements with G.813 clocks. In contrast with the PSTN, only one type of G.812 slave clock is shown because the difference in holdover performance of the TNC and LNC is not relevant for SDH network synchronization. For simplification, the reference chain in Figure 5.13 does not exhibit the distribution facility of the synchronization signal. Before now, practical measurements have not been done to verify this "ideal" reference model. The influence of time jitter and wander, affected by the transport network, have not been considered yet in this reference connection. Nevertheless, clock noise here is negligible in comparison with that of the transmission line as the length of the reverence chain may be shortened in practice.

So, the new trends in network synchronization are based on distributed PRCs with shorter synchronization chains. The slave clocks have more than one input for the reference signals received from different PRCs. For reliable and stable operation of such synchronization networks, perfect network management is required. The ITU-T Recommendation G.781 contains a detailed description of synchronization trails' organization with correspondent Quality Levels of synchronization signals. For example, Option I of SDH synchronization networking (for 2 048 kbit/s hierarchy) has 4 levels of synchronization quality (Table 5.3).

Table 5.3 – Synchronization quality levels

Quality Level [G.781, Option I]	The clock type generating synchronization signal with corresponding QL
QL-PRC	G.811 (09/97) Primary Reference Clock
QL-SSU-A	G.812 (09/97) Type I (or Type V Slave Clock that equals G.812 (11/88) TNC)
QL-SSU-B	G.812 (09/97) Type IV Slave Clock equals G.812 (11/88) TNC
QL-SEC	G.813 (08/96) Option I SEC
QL-DNU	This signal should not be used for synchronization

The ITU-T Recommendation G.812 deals with three main types of SSU:

Type I is for 2 048 kbit/s networking (the wander generation and bandwith of Type I clocks are limited to the deployment in the reference synchronization chain in Figure 5.13);

Type II is for distribution hubs – 1 544 kbit/s networking;

Type III is for end offices – 1 544 kbit/s networking.

Moreover, the ITU-T Recommendation G.812 includes three additional types of SSU for existing networks:

Type IV must be complied with G.813, Option II (1 544 kbit/s networking);

Type V equals TNC of G.812 (11/88);

Type VI equals LNC of G.812 (11/88).

5.6 Synchronization in ATM

The main feature of Asynchronous Transfer Mode (ATM) is that data transfer within the network itself is asynchronous but it is not related to the service. An ATM network must handle any traffic type including those which are inherently timing dependent. So the synchronization of the network is one of the most important issues in the design of the ATM network. There are some traffic types that require synchronization between source and destination such as [17]:

– Constant-bit-rate (CBR) services;

– Voice (including compressed);

– Video (including compressed);

– Multimedia.

Only ATM networks that are not based on these service types (data networks with variable-bit-rate (VBR), for instance) have no need for synchronization.

5.7 References

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5.8 Abbreviations

ADC	Analogue-to-digital Converter
ADSL	Asymmetric Digital Subscriber Line
ANSI	American National Standards Institute
ATM	Asynchronous Transfer Mode
BP	Band Pass (Filter)
BSRF	Basic Synchronization Reference Frequency
CCITT	International Telephone and Telegraph Consultative Committee (predecessor to the ITU-T)
CBR	Constant-Bit-Rate
CPE	Customer Premises Equipment
DA	Data-aided (synchronization)
NDA	Non-data-aided (synchronization)
E1	European digital telephony format (given to the Conference of European Postal and Telecommunication Administration) that carries data at the rate of 2.048 Mbit/s
4ESS	No. 4 Electronic Switching System – Bell System's tool and tandem system based on digital time division switching
GPS	Global Positioning System (USA)
GLONASS	GLObal NAvigation Satellite System (Russia)
IEEE	Institute of Electrical and Electronic Engineers
ITU-T	International Telecommunications Union – Telecommunications (formerly CCITT)

LP	Low Pass (Filter)
LNC	Local Node Clock
ML	Maximum Likelihood
PLL	Phase Locked Loop
PCM	Pulse Code Modulation
POTS	Plain Old Telephone Service
PRC	Primary Reference Clock
PSTN	Public Switching Telephone Network
SEC	SDH Equipment Slave Clocks
SDH	Synchronous Digital Hierarchy
SONET	Synchronous Optical NETwork
SSU	Synchronization Supply Unit
TEC	Timing Error Corrector
TED	Timing Error Detector
TNC	Transit Node Clock
UTC	Coordinated Universal Time
VBR	Variable-Bit-Rate
VCXO	Voltage Controlled Crystal Oscillator
VDSL	Very-high-rate Digital Subscriber Line
XO	Crystal Oscillator

CHAPTER 6

6 Digital transmission (PDH, SDH, DWDM, xDSL)

6.1 Plesiochronous digital hierarchy (PDH)

6.1.1 Principles

In the beginning of the 1960s new transmission techniques were established using Pulse Code Modulation, PCM. These techniques are a combination of pulse code modulation and time division multiplex transmission and, like FDM carrier techniques, permit utilisation of a single transmission circuit for more than one telephone channel.

The basic principle of PCM transmission involves the periodic sampling of voice-frequency signals. The sampling frequency is plesiochronous, i.e. a nominal rate with a permissible deviation from this rate. The analogue amplitude obtained at each sampling instant is converted into a number of digital pulses arranged within one time slot. The signal transmitted to line consists of a series of such time slots which, at the receiving terminal, are reconverted into a series of analogue signals, permitting the reconstruction of the speech information in analogue form. The speech time slots are arranged in frames containing time slots for frame synchronization and signalling.

PCM line signals consist of digital pulses, which can be regenerated by relatively simple digital repeaters placed at regular intervals. Signal-to noise ratio, attenuation and distortion are almost independent of the number of repeaters. Limitations occur due to accumulated jitter. The less stringent demands on the quality of the transmission medium are gained at the cost of increased bandwidth requirements. The PCM coding-decoding process, where an infinite number of amplitudes are represented by a finite number of quantified samples leads to quantization noise, which can be reduced by non-linear coding methods.

Among the advantages of digital transmission is the capability of encompassing all types of signals which are represented in digital form: e.g. speech, data, sound and video.

6.1.2 Standards

ITU has defined PCM – and Digital Multiplexers in its G-Series of Recommendations.

Some of the more important Recommendations are listed below with abbreviated titles:

– Digital Hierarchy bit rates (G.702);

– Characteristics of digital interfaces (G.703);

– Principal characteristics of multiplex equipment (G.731-G.755).

Three multiplex hierarchies exist worldwide, differing in bit rates, frame format and coding methods, see Table 6.1.

Table 6.1 – PDH levels

PDH Level	Europe (A-law)		USA (μ-law)		Japan (μ-law)	
0	64 kbit/s	1 ch	64 bit/s	1 ch	64 kbit/s	1 ch
1	2048 kbit/s	30 ch	1544 kbit/s	24 ch	1544 kbit/s	24 ch
2	8448 kbit/s	120 ch	6312 kbit/s		6312 kbit/s	
3	34.368 Mbit/s	480 ch	44.736 Mbit/s		32.064 Mbit/s	
4	139.264 Mbit/s	1920 ch	139.264 Mbit/s		97.728 Mbit/s	

The Recommendations describe the functions of the equipment, but not implementation details. Some of the essential functions are:

Sampling	The amplitude of the 4 kHz bandwidth analogue signal is measured (sampled) 8000 times per second.
Quantization and Coding	Each sample is represented by an 8 bit code permitting 256 code words which represent 256 analogue signal levels.
Channel	Each telephone channel is represented by 8 bit words repeated 8000 times per second leading to a bit rate of 64 kbit/s.
Framing	The timeslots representing various channels are multiplexed together forming a frame, which is identified by a frame alignment word. In addition frames can contain signalling and management information.
Management	The management bits in frames lead to defined actions and consequent signals, simplifying the identification of the source and location of faults.

6.1.3 Implementation

Figure 6.1 illustrates plesiochronous transmission from a pcm multiplexer to an exchange using digital multiplexers and digital line systems. Each digital multiplexer has its own timing source. Due to bit interleaving at each multiplexer level the 8 bit (byte) structure of each channel is only accessible at the end points of the connection. The different multiplexer timing sources make it impossible to retrieve a particular channel (time slot) from a higher order frame without additional multiplexing / demultiplexing equipment. In addition each plesiochronous frame provides very little overhead information which would be required for efficient network management. In spite of the earlier described PDH advantages compared to analogue transmission, plesiochronous transmission is not appropriate for future integrated networks.

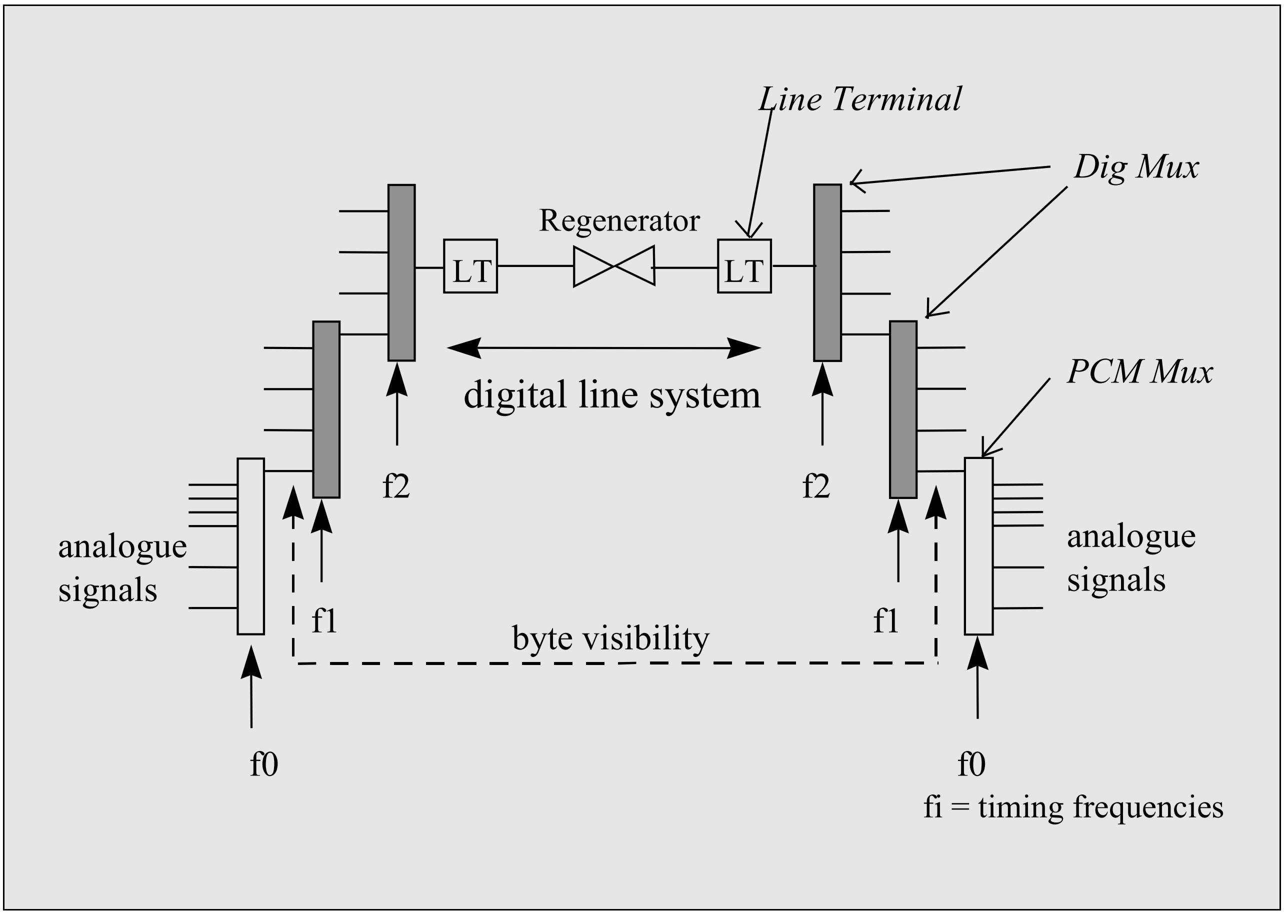
Plesiochronous systems are used e.g.:

– in rural areas with overhead pair cables, (multiplex levels 1 and 2)s;

– inlocal areas for connection of radio stations (multiplex levels 1 and 2);

– in long distance network with coaxial and fibre cables (multiplex levels 3, 4 and 5).

Figure 6.1 – Example of plesiochronous multiplexing



6.2 Synchronous digital hierarchy (SDH)

6.2.1 Principles

By the middle of the 1980s the increased demand for higher transmission bit rates, more flexible channel handling, together with more elaborate management requirements, lead to the concept of the Synchronous Digital Hierarchy. SDH extends the principles of PDH by defining new administrative levels derived by direct byte interleaving, leading to a number of advantages:

- Digital 64 kbit/s channels or groups of channels can be added to or extracted directly from SDH signals leading to economic ADD/DROP equipment.
- Plesiochronous signals of different levels and belonging to different hierarchies (e.g. CEPT, ANSI) can be mapped to SDH and transmitted over SDH signals.
- Digital channels belonging to different levels can be switched in Synchronous Digital Cross Connect equipment (DXC).
- The routing in DXC networks can be command controlled permitting, in a flexible way, different logical network configurations at different times.

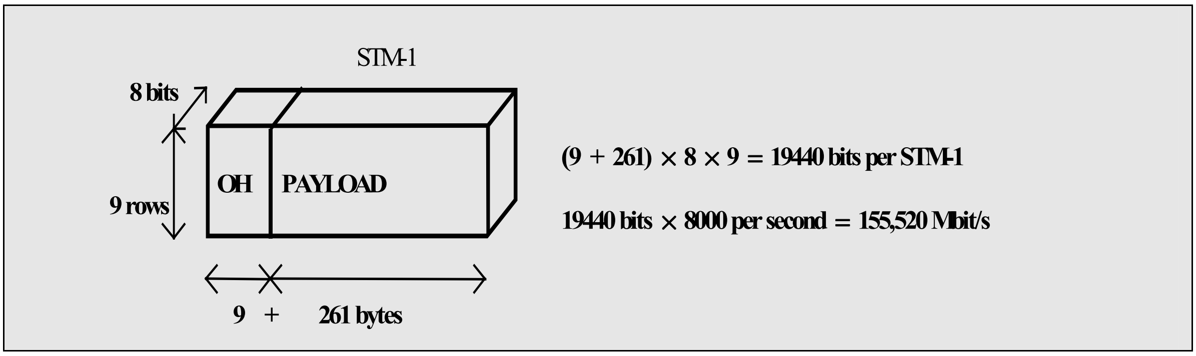
- DXCs permit sorting (e.g. the separation of data, voice and video signals) and packing (economic utilisation of the transmission media) of the transmitted information.

- Last but not least, SDH and DXC equipment has been designed for network management based on the principles of Telecommunication Management Networks (TMN).

SDH signals are transmitted in the form of Synchronous Transport Modules (STM). The module STM-1, as shown in Figure 6.2, contains 2349 data bytes (STM payload) and 81 overhead bytes (STM OH) which are transmitted 8000 times per second, leading to the transmission bit rate 155.520 Mbit/s. Higher transmission rates $155.52 \times N$ are achieved with the synchronous transport modules STM-N with SDH levels $N = 4$, 16 and 64.

The STM payload contains information according to the SDH multiplexing scheme illustrated in Figure 6.3. Plesiochronous signals are converted (mapped) to synchronous signals, they are inserted in Virtual Containers (VC), multiplexed and transmitted as Synchronous Transport Modules (STM) across a digital line. VCs and STMs contain in addition, ample overhead information for the management and phase alignment of the digital signals in the various multiplexing stages.

Figure 6.2 – Illustration of synchronous transport module STM-N



6.2.2 Standards

The ITU has defined SDH Equipment and its management in the G Series of Recommendations. Some of the more important Recommendations are listed with abbreviated titles and explained in more details below:

– Digital networks and their architecture (G.801, G.802, G.803, G.805);

– Principal characteristics of multiplexing equipment for SDH (G.781, G.782, G.783);

– Digital Line Systems (G.957, G.958);

– Synchronous Digital Hierarchy Management (G.773, G.774 and G.784).

The Synchronous Optical Network (SONET) is based on 52 Mbit/s (USA standard), which can be multiplexed to STM-1.

Table 6.2 below shows SDH levels, related bit-rates and number of 64 kbit/s channels:

Table 6.2 – SDH levels

SDH levels	Bit rate (kbit/s)	No. of telephone channels
STM-1	155,520	1,920
STM-4	622,080	7,680
STM-16	2,488,320	30,720
STM-64	9,953,280	122,880

Figure 6.3 shows the world-wide accepted multiplexing schemes and Figure 6.4 illustrates a multiplexing example based on this scheme.

A plesiochronous 2 Mbit/s tributary signal is mapped into a synchronous Container (C-12) with the addition of justification bits for frequency adaptation. After addition of Path Overhead information (VC-12 POH) for path management the Virtual Container (VC-12) is obtained. The phase difference between VC-12 and Tributary Units (TU) is indicated by a Tributary Unit Pointer, after addition of the pointer the Tributary Unit TU-12 is obtained (pointer processing). In the first multiplexing stage three TU-12s are inserted into one Tributary Unit Group (TUG-2). The second multiplexing stage combines 7 TUG-2s and Path Overhead information (VC-3 POH) in the Virtual Container VC-3. After addition of pointer information for phase alignment the Administrative Unit AU-3 is obtained (pointer processing). The third multiplexing stage multiplexes 3 AU-3s to the Administrative Unit Group (AUG). Finally in the forth multiplexing stage N AUGs are byte interleaved. STM-N is obtained after addition of Repeater Section Overhead (RSOH) for the management of repeater sections and Multiplex Section Overhead (MSOH) for the management of multiplex sections.

In such a way a STM-1 can transmit $3 \times 7 \times 3 = 63$ 2 Mbit/s tributaries or 3×34 Mbit/s tributaries or one 140 Mbit/s tributary.

Figure 6.3 – SDH multiplexing scheme

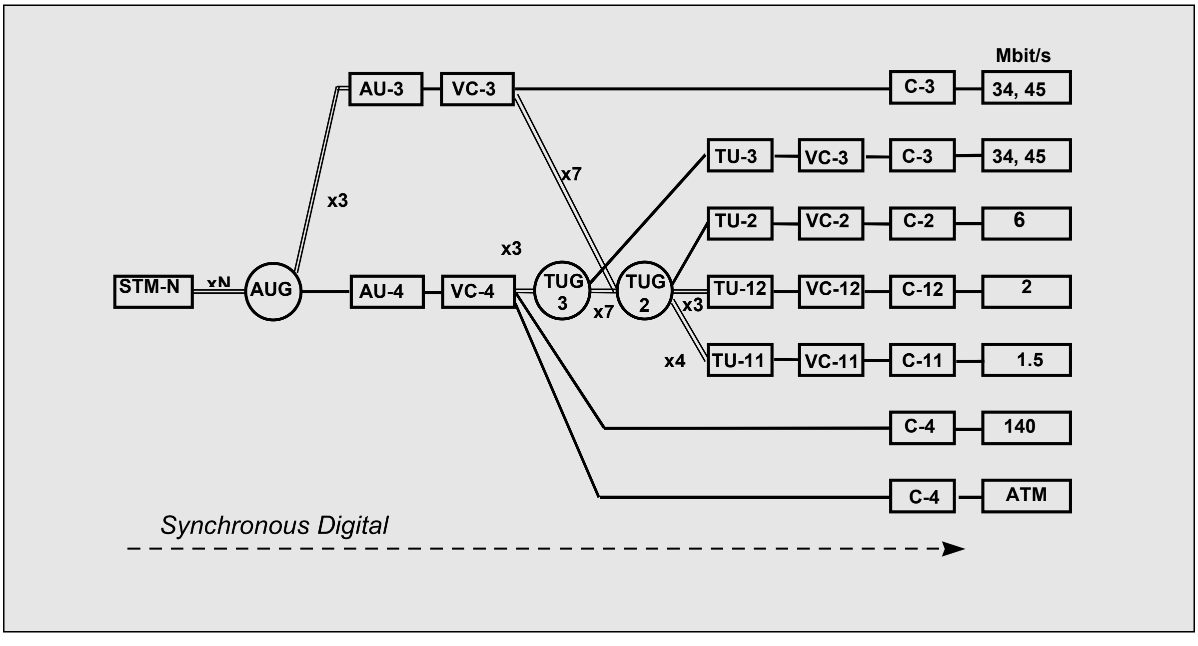
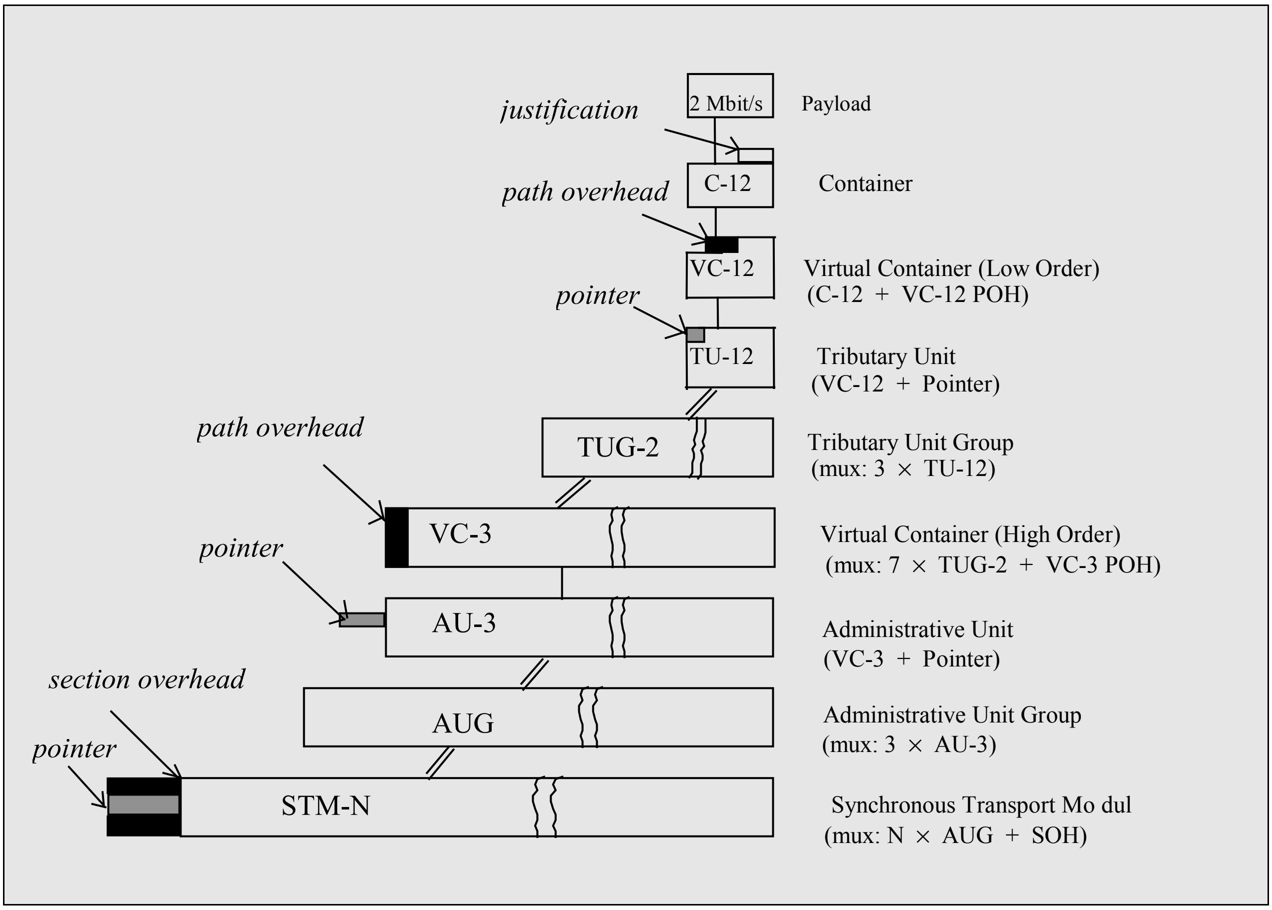


Figure 6.4 – Mbit/s multiplexing example

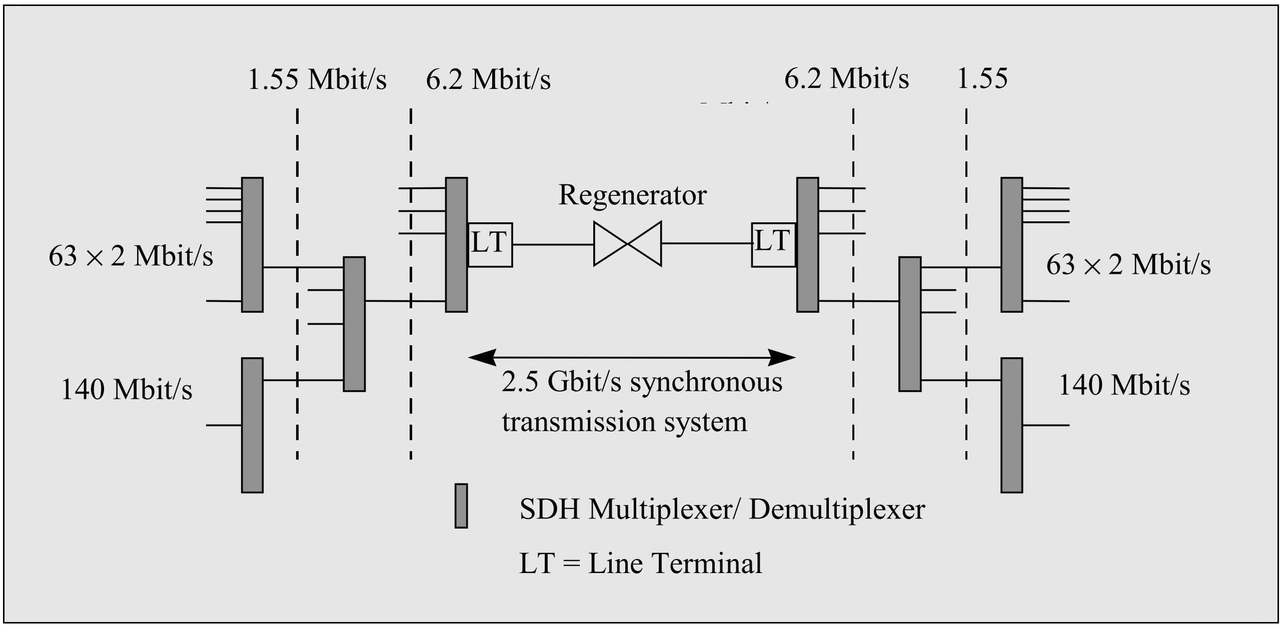


The SDH Recommendations describe in detail the overhead information offered by STM-N for management of VC paths, Multiplexer, and Regenerator Sections and the extensive management functions provided across the TMN Q3 interfaces. The SDH terminals are equipped with Multiplex Section Protection for the protection of point-to-point connections. Multiplex Section Shared Protection Rings (MSSP), consisting of 2 bi-directional rings connected to Add/Drop multiplexers, offer efficient protection against node or ring failures.

6.2.3 Implementation

SDH networks contain SDH Multiplexers (Terminal and ADD/DROP Multiplexers), Digital Cross Connects (with different cross connect levels) and Digital Line Systems (using coaxial or fibre cables). Figure 6.5 illustrates a typical SDH transmission network for the transmission of plesiochronous tributaries.

Figure 6.5 – Example of SDH transmission network



A number of multiplexer types has been standardized and are shown in Figure 6.6:

Type I for conversion from plesiochronous to synchronous STM-N signals.Signals between plesiochronous terminals or network parts can be transmitted using the benefits of synchronous transmission.

Type II for conversion between various STM signals. A number of STM-1 signals can be multiplexed to a higher bit rate to use fibre optic cables more efficiently.

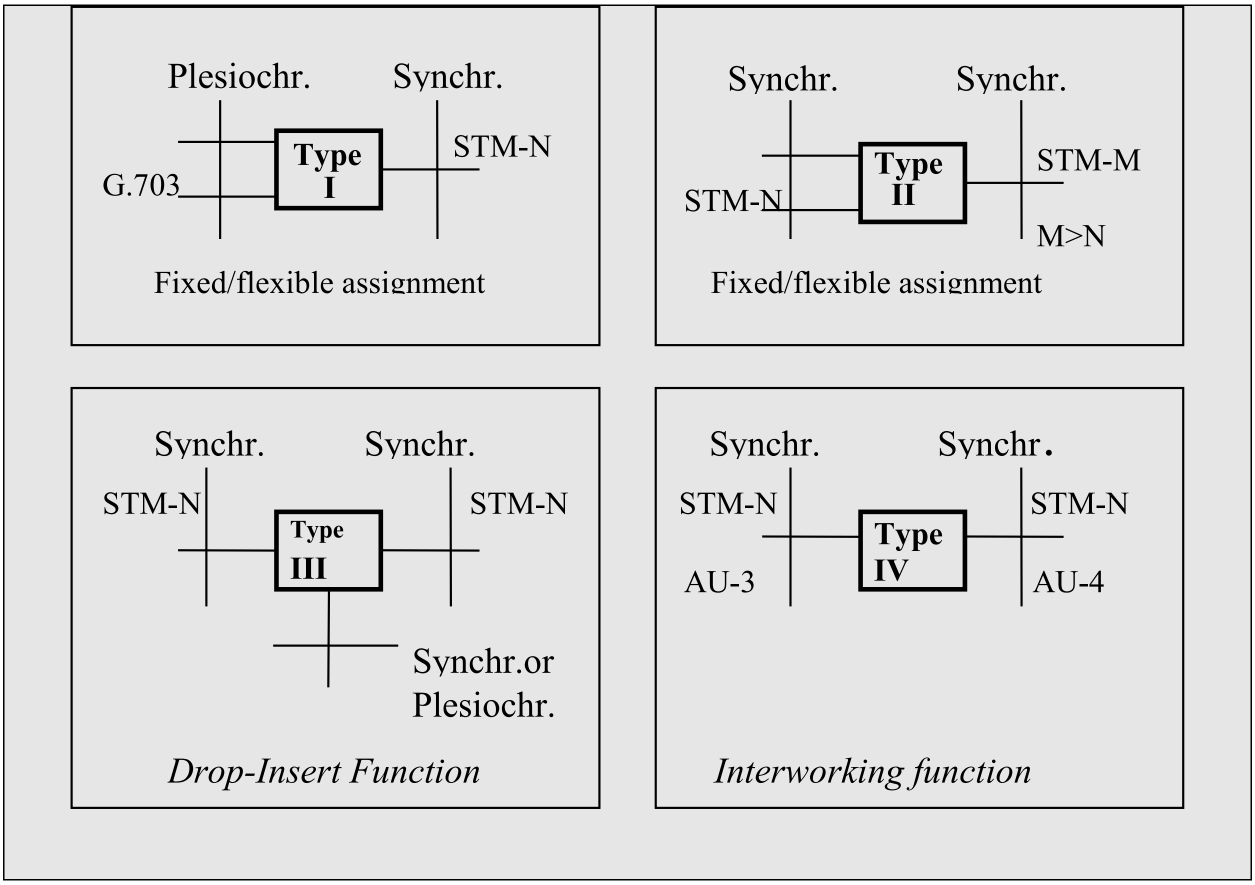
Type III for drop/insert of plesiochronous and synchronous signals to STM-N. Single channels or groups of channels can be added or dropped from a synchronous bit flow, e.g. for Add/Drop multiplexers in ring configurations.

Type IV for interworking between AU-4 (CEPT) and AU-3 (ANSI). Interworking between different hierarchies e.g. if CEPT related SDH signals have to be transmitted across ANSI facilities and vice versa.

Types I and II permit flexible channel assignment, which corresponds to a limited cross connect functionality.

Digital Cross Connect equipments permit the switching of various synchronous and, after mapping, plesiochronous tributaries which enter and exit the DXC at the port level. The level which is used for the switching (cross connect level) is equal or less than the port level. Typical port levels are: 2, 34, 140 Mbit/s and STM-1 signals. Typical cross connect levels are: VC-12, VC-3, VC-4.

Figure 6.6 – SDH multiplexer types



The various Digital Cross Connect equipments can be characterized by these levels as illustrated below:

DXC 1/0 e.g. port level 2.048 Mbit/s and cross connect level 64 kbit/s for 64 kbit/s leased line networks;

DXC 4/1 e.g. port level 140 Mbit/s and cross connect level VC-12 for 2 Mbit/s leased line networks;

DXC 4/4 e.g. port level 140 Mbit/s or STM-1 and cross connect level VC-4 for network protection, together with DXC 4/1 for network administration.

Depending on the requirements, a number of DXC combinations have been considered, e.g.

Table 6.3 – DCX types

DXC Type	Plesiochronous and synchronous port levels (and cross connect levels)
Type I	140 Mbit/s/STM-N (VC-4)
Type II	2, 34, 140 Mbit/s/STM-N (VC-12)
Type III	2, 34, 140 Mbit/s (VC-12) and 34 Mbit/s (VC-3) and 140 Mbit/s (VC-4)/STM-N

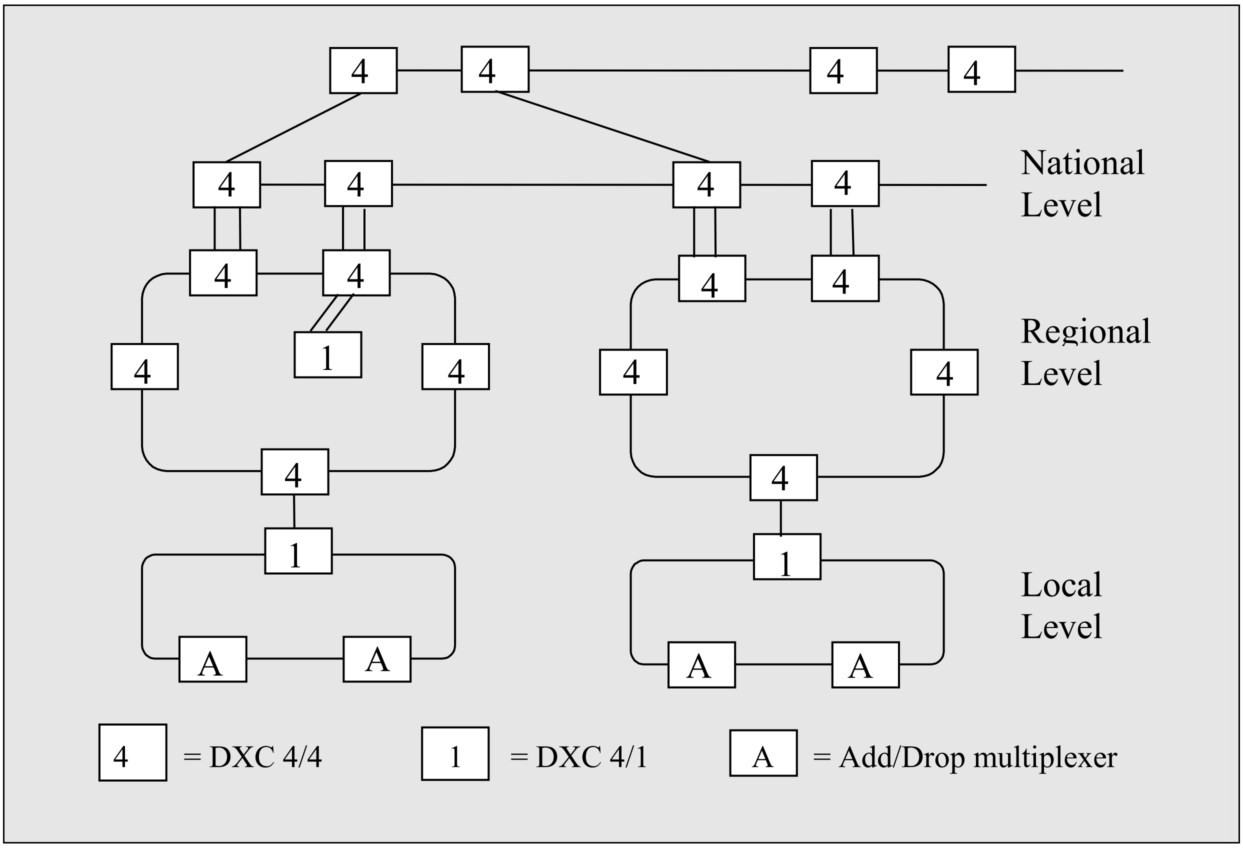
Digital line systems, SDH multiplexers and DXC equipment are the ingredients of new digital transmission networks. A typical configuration is shown in Figure 6.7. The network in the example consists of 3 network layers (national, regional and local network layers) using mainly ring configurations in each layer. The reasons behind this network structure are:

– the flexibility of DXCs increases the degree of utilisation and simplifies planning, e.g. connections between customers in different access network parts use the regional layer and the national layer for the establishment of the connection;

– MSSP rings, and the use of 2 node connections between 2 adjacent rings, offer a very high level of reliability;

– the simple network structure is suitable for TMN, which simplifies operation, administration, and maintenance, decreasing management costs.

Table 6.7 – SDH network example



The introduction of SDH can follow three different strategies:

- The top-down strategy in which SDH is first employed in the trunk and long distance network to offer increased transmission capacity with improved management capability.
- The bottom-up strategy in which SDH is first introduced in the local network forming SDH islands, e.g. for local customers requiring high speed data interconnection.
- The overlay strategy which leads to separate SDH networks, e.g. for distributed customers requiring improved quality and data rates compared with existing networks.

6.3 Dense Wavelength Division Multiplex (WDM)

6.3.1 Principles

The continuous demand for increased bandwidth at reasonable cost, together with the evolution of fibres and fibre optical components, led to the renascence of frequency division systems (FDM). However, as the copper-bound FDM systems could only transmit about 10000 analogue telephone channels, high density wavelength division multiplex systems (DWDM) on fibres will be able to transmit about 30 Million digital telephone channels or a corresponding number of broadband channels.

Figure 6.8 illustrates the evolution of fibre optical transmission systems.

In the early 1970s systems with light emitting diode transmitters (LED) using multi-mode fibres in the 1.3 µm band and 10 km repeater distances could transmit up to about 300 Mbit/s.

During the late 1980s systems with Laser transmitters using single-mode fibres in the 1.55 µm band and 100 km repeater distances could transmit up to about 2500 Mbit/s.

The late 1990s systems with parallel working, narrow-band lasers at different frequencies and Erbium-doped fibre amplifiers using single-mode fibres in the 1.55 µm band and 120 km amplifier distances will transmit up to about 2500 Gbit/s. However, the total distance between DWDM terminals will be limited to about 600 km.

Figure 6.8 – Evolution of optical fibre transmission systems

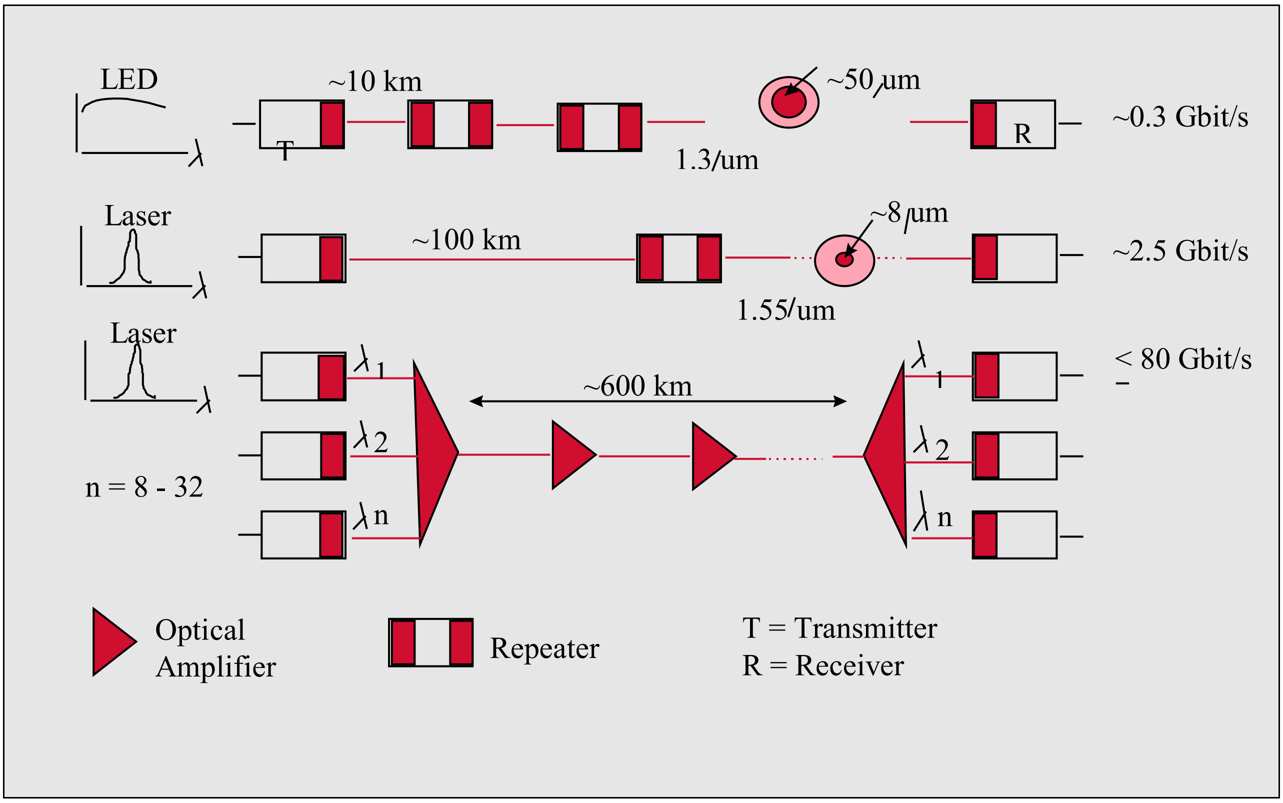


Table 6.4 – DWDM advantages and disadvantages

DWDM Advantages	DWDM Disadvantages
DWDM offers high transmission rates on fibre pairs, the capacity can be increased without laying new cables as existing cables can be used. The transmission capacity can be increased in a modular manner by adding additional wavelengths as and when capacity increases are required. DWDM systems are transparent, i.e. different wavelengths can carry different data at different data rates. Two-way DWDM transmission is possible across a single fibre.	Due to imperfections in the present optical amplifiers the number of amplifiers and the total distance between DWDM terminals is limited. DWDM systems offer less monitoring and management capabilities compared with time division multiplex systems.

6.3.2 Standards

ITU has defined optical networking in its G Series of Recommendations.

Some Recommendations with abbreviated titles are shown below:

– Optical line systems with optical amplifiers and optical multiplexing (G.681).

– Optical interfaces for multi-channel systems with optical amplifiers (G.692).

– Structures and mapping for the optical transport network (G.709).

– Functional characteristics of optical networking equipment (G.798).

– Framework for optical networking Recommendations:

– Architecture, Requirements and Management (G.871, 872, 873, 874, 875).

The Recommendations describe

– frequency allocations,

– access and line interfaces,

– management functions.

Figure 6.9 shows the recommended ITU wavelength grid, permitting up to 43 optical frequencies in the 1.55 µm window.

An additional frequency is employed as Optical Supervisory Channel (OSC). Figure 6.10 shows as an example an OSC arranged at 1.51 µm, i.e. outside the frequency band of the payload channels. The OSC can carry a 2 Mbit/s framed signal permitting it to transmit ample management information e.g. optical section trace and status, laser output power control, data communication channels and engineering orderwire.

Figure 6.9 – ITU wavelength grid

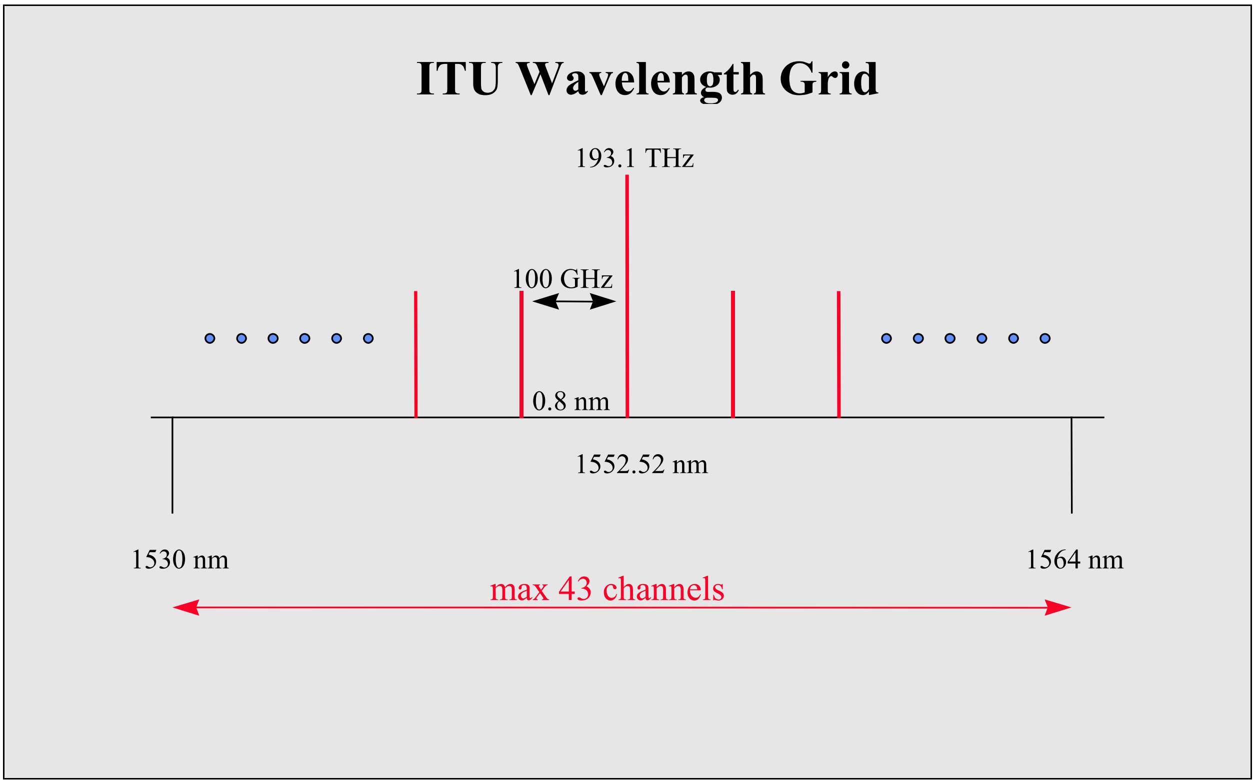
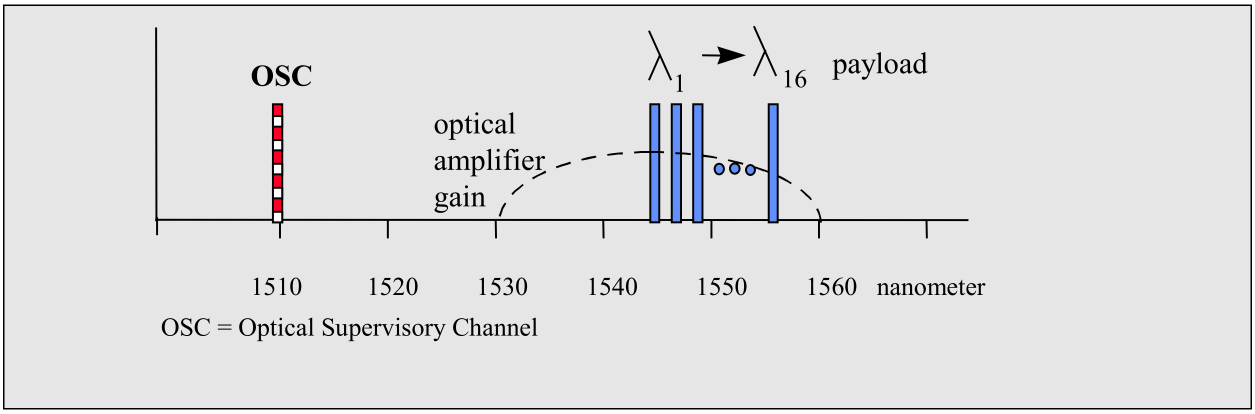


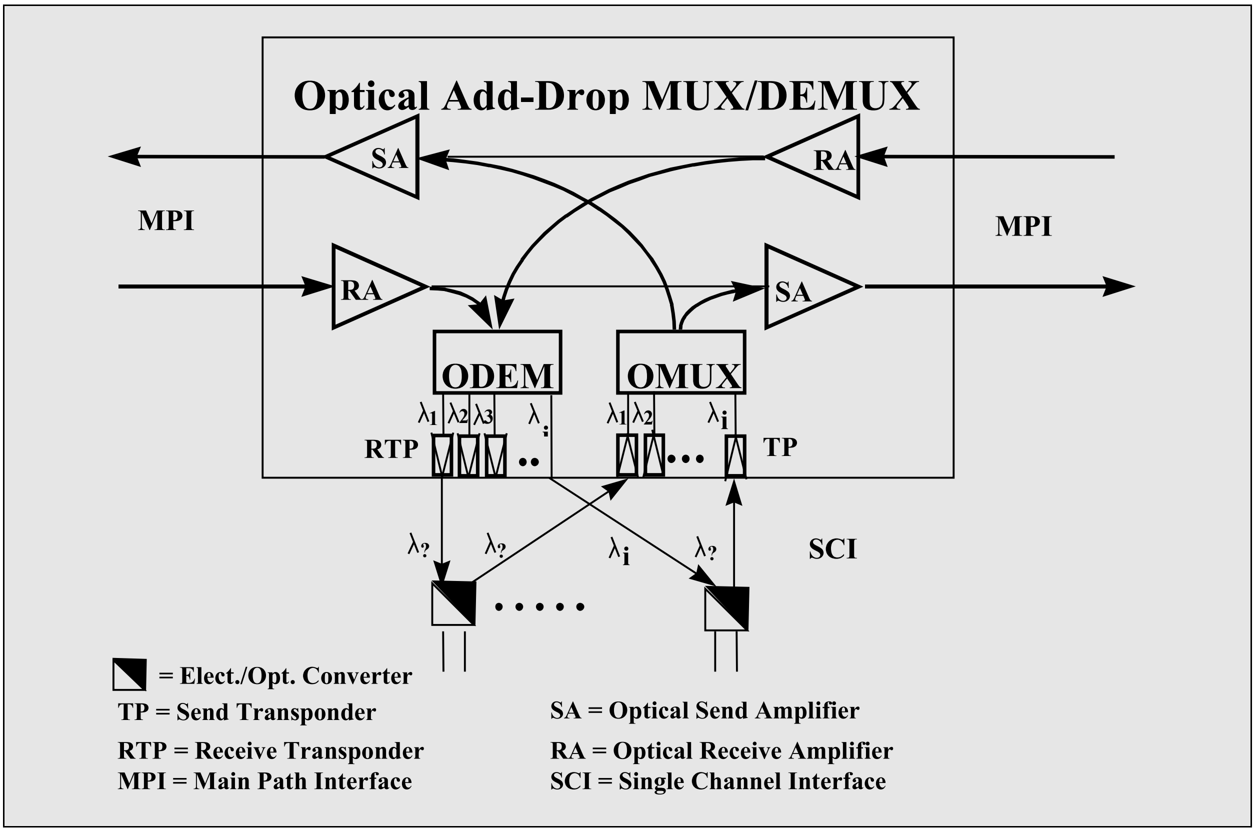
Figure 6.10 – Optical supervisory channel



6.3.3 Implementation

Figure 6.11 illustrates optical multiplexing/demultiplexing and add-drop functionalities in an Optical Add-Drop Multiplexer/Demultiplexer (OADM). Incoming optical tributaries at the Single Channel Interface (SCI) normally have unspecified wavelengths generated in Electrical/Optical Converters. They are converted to standardized wavelengths in a send transponder (TP), multiplexed in a fibre coupler assembly and transmitted via an optical send amplifier (SA) to the Main Path Interface (MPI). In the receive direction incoming line signals are amplified in an optical receive amplifier (RA) and demultiplexed in discrete wavelength filters, e.g. in planar-array waveguides. Receive transponders (RTP) are required if 1.55 µm signals are not suitable for the subsequent converters. In addition to the add/drop frequencies other frequencies can pass through the OADM. The add-drop function of OSC is not shown in the Figure.

Figure 6.11 – Optical add-drop multiplexer/demultiplexer example



Figures 6.12 and 6.13 show typical optical network configurations.

The Point-to Point network consists of Optical Terminal Multiplexers/Demultiplexers (OTM) and Optical Line Amplifiers (OLA). For management the OSC is connected via a Control Unit (CON) and a Q3 management interface to a Work Station (WS).

With Optical Add/Drop Multiplexers/Demultiplexers (OADM) Optical Point-to-Point protection networks are feasible. The switch-over from working to protection section is initiated by signal detection and activated by OSC functions.

A number of OADMs can be arranged in rings, offering flexible network and traffic configurations. In the case of node or line faults re-routing in the ring can be activated by OSC functions. In the example the traffic between the nodes A + B, A + C and C + D uses the frequency λ1, and the traffic between nodes A + D is based on the frequency λ2. A fault between nodes A and C results in re-routing via nodes B and D.

Figure 6.12 – Optical supervisory channel

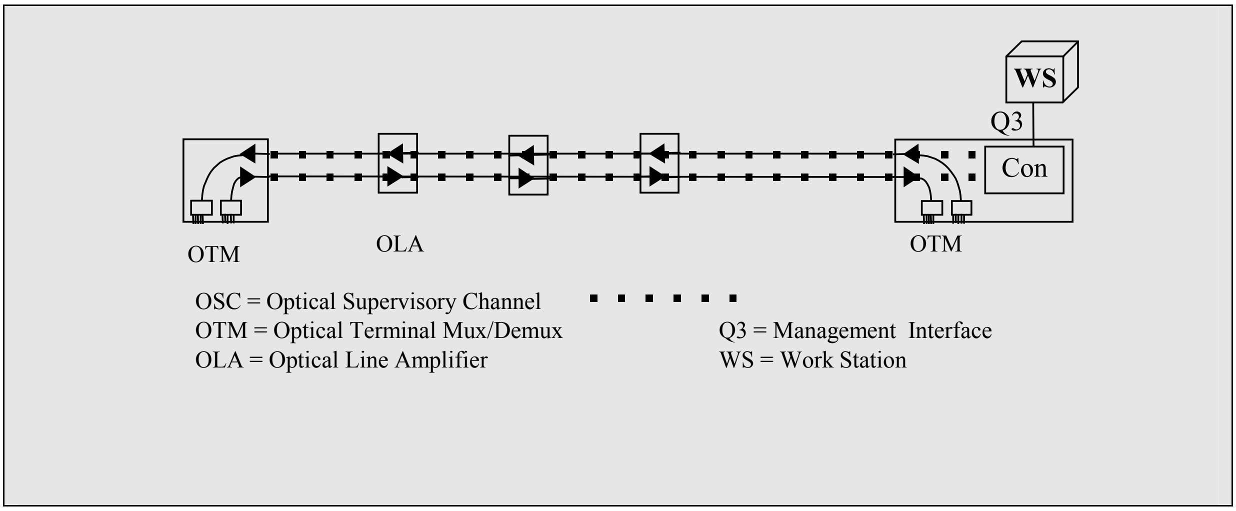
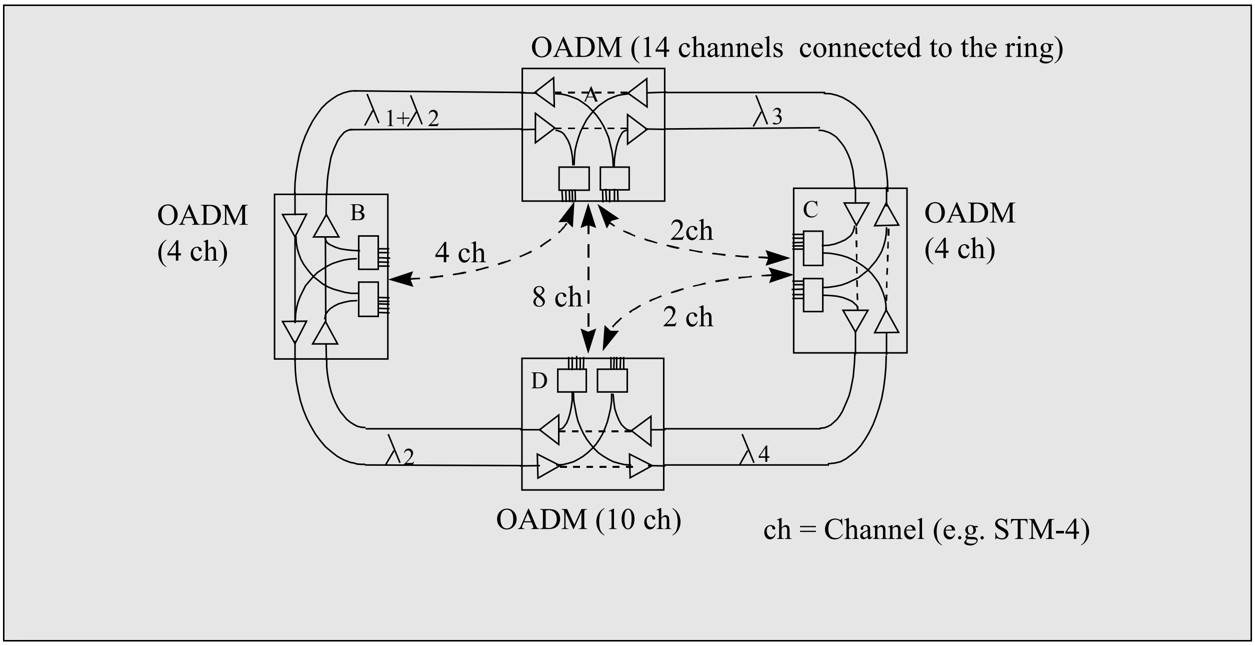
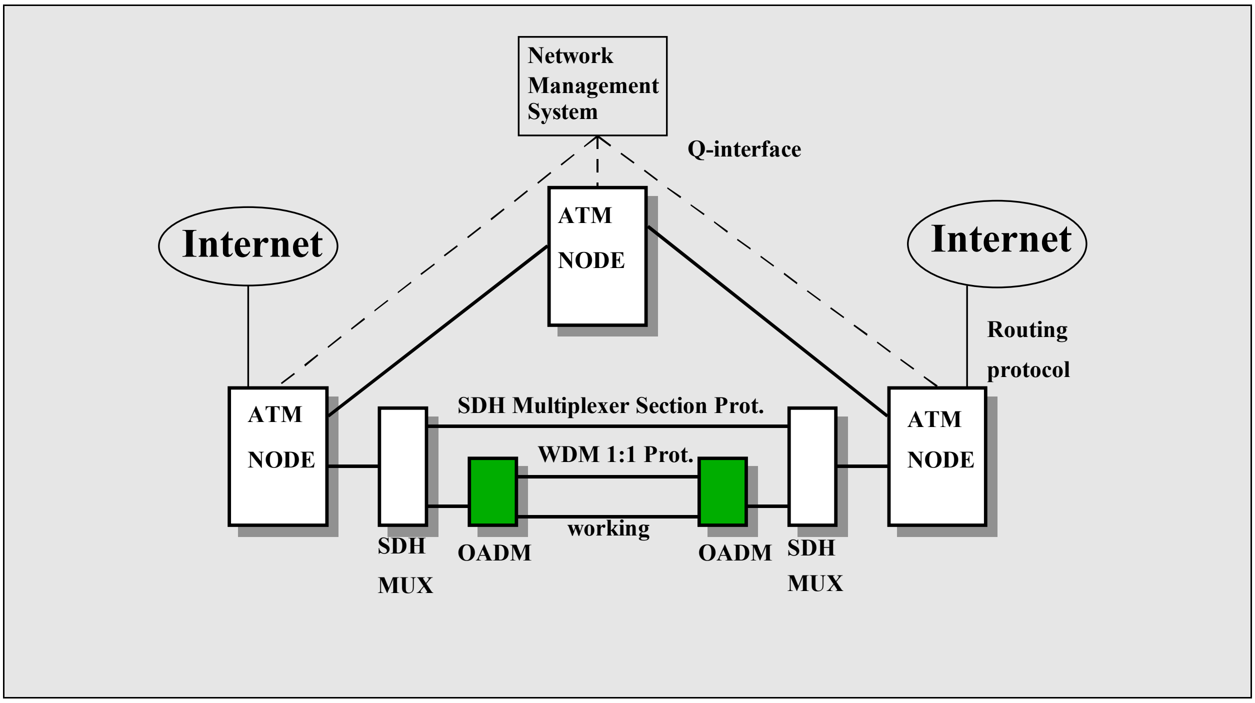


Figure 6.13 – Optical ring network traffic example



Future transmission networks are expected to be based on ATM for the establishment of services. ATM signals can be carried by SDH transmission and SDH signals can be carried by DWDM transmission. As illustrated in Figure 6.14 the combination of these technologies permits a variety of different protection and routing schemes. The example shows DWDM and SDH protection which co-operates with the re-routing facilities of ATM nodes, e.g. controlled by a network management system. In the case of Internet connections additional routing is possible using Internet routing protocols. Proper choice of protection mechanisms is necessary to guarantee the high performance requirements of existing and new services.

Figure 6.14 – Network protection



6.4 Digital subscriber lines (XDSL)

6.4.1 Subscriber Loops

Users of telecommunication services, subscribers, are connected across the access network to the transit networks. Historically these connections, subscriber loops, consist of twisted copper pairs assembled in cables. Figure 6.15 shows an example of telephone loop plant with feeder cables to concentrated customer areas, distribution cables to potential customer sites and drop wires to customer premise.

Subscriber loops have been under study for many years and different models exist to describe important parameters such as:

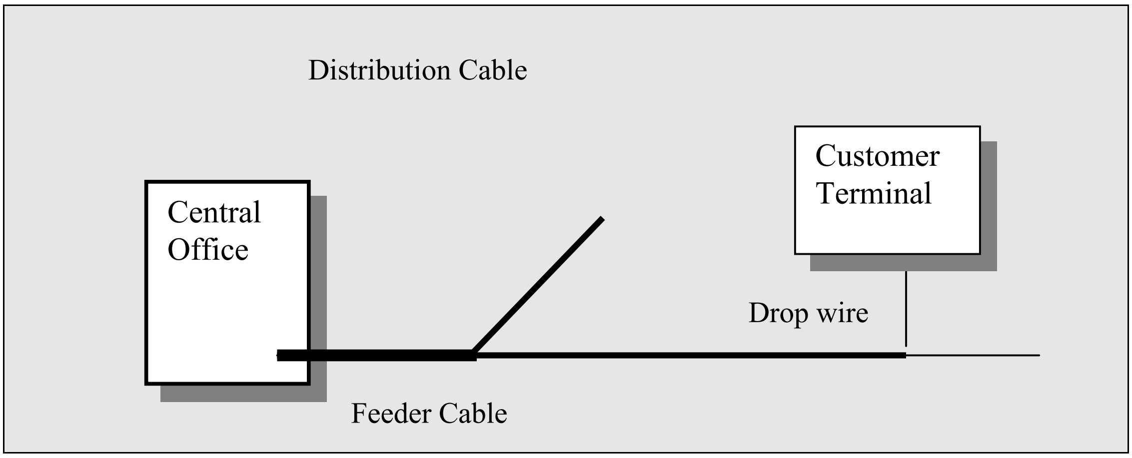
– cable type (wire diameter, isolation material);

– cable length;

– loop structure (load coils, bridged taps);

– noise sources (cross talk, impulse noise, radio frequency interference).

Figure 6.15 – Example of telephone loop plant



For analogue voice frequency signals, it is normally the attenuation, based on the wire gauge, determines the length of a subscriber loop. Loading coils are used in some cases to extend the range.

For digital signals with bandwidths exceeding voice frequencies it is normally attenuation, crosstalk and phase delay which limit subscriber loop lengths. In addition impulse noise can influence the range.

The introduction of new services demanding digital signals with higher and higher bit rates made it necessary to either extend the usable bandwidth of existing subscriber loops with sophisticated technologies, or to replace the twisted pairs with broadband transmission media such as fibre/coaxial cables or wireless transmission.

The extensive cost involved in replacing the existing subscriber loops, together with developments in the field of digital signal processing, led to the development of new technologies for better utilization of the available bandwidth such as:

– baseband modems;

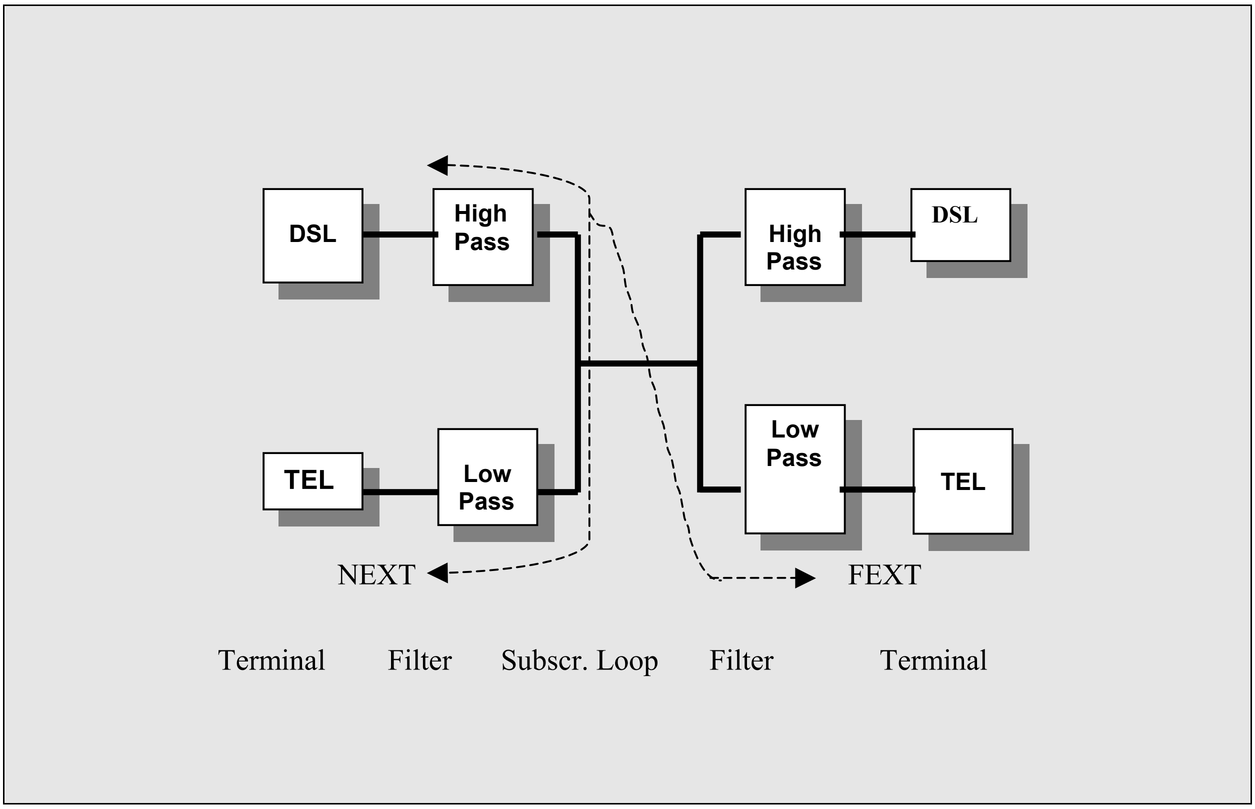
– voice band modems;

– digital subscriber lines (xDSL).

About 15% of existing subscriber loops require upgrade activities to be suitable for xDSL techniques. Corrective measures include the introduction of mid-span repeaters and the removal of load coils, and in certain cases the removal of bridged taps. Simultaneous transmission of voice frequency signals and higher frequency signals, in the same or opposite directions, can require splitter installation as illustrated in Figure 6.16.

Near-end crosstalk (NEXT) is a major impairment for systems that share the same frequency band for upstream and downstream transmission. NEXT noise is seen by the receiver located at the same end of the cable as the transmitter which is the noise source. Far-end crosstalk (FEXT) is the noise detected by the receiver located at the far end of the cable from the transmitter which is the noise source. FEXT is less severe than NEXT because the FEXT noise is attenuated by traversing the full length of the cable.

Figure 6.16 – Example of splitter installation



Splitter filter configurations make it possible to isolate POTS and DSL applications. Splitters decrease the influence of on-off hook related impedance changes, pulse, ringing and crosstalk disturbances. Near-end crosstalk has to be attenuated as a DSL transmitter sends with about 100 mW and a telephone receiver works with 0.1 mW.

The usable spectrum of the twisted pair can be extended from the voice frequency signals of up to 4 kHz, to over 500 kHz for the transmission of digital signals using xDSL technologies. With powerful digital signal processing it is possible to use sophisticated coding methods which mitigate crosstalk. Together with channel equalization and echo cancellation techniques bit rates in the range of Mbit/s can be transmitted across existing physical subscriber loops. Subscriber Line Systems are designed with a 6 dB signal to noise margin to secure a bit error rate better than 10 exp –7. The design margin provides e.g. for cable variations and noise impairment and is a trade of between reliable operation and transmission across the longest possible loop.

6.4.2 Subscriber Line Systems

Voice band modems were introduced in the late 1950s to transmit data in the voice frequency band from 200 Hz to 3400 kHz. More and more bandwidth efficient modems were developed, in late 1996 the technological limit for modems were achieved with the standardization of the ITU V.90 modems offering up to 56 kbit/s transmission across a dial-up telephone connection with a bandwidth efficiency of about 14 bits/s per Hz.

Table 6.5 below relates the existing and new technologies to access bit rates and applications:

Table 6.5 – Subscriber line technologies

Type	Description	Access/Bit rates/range	Applications
BB	Baseband Modems	Duplex: 32 kbit/s to 2 Mbit/s Range: few km/1 pair	Leased line connection.
V.22 to V.90	Voiceband Modem connections	Duplex: 1.2 kbit/s to 56 kbit/s Range: unlimited/1 pair	Dial-up connection.
DSL	Digital Subscriber Line	Duplex: 160 kbit/s (2B + D + MI) Range: up to 5.5 km/1 pair	ISDN service.
HDSL	High Data-rate Digital Subscriber Line	Duplex: 2 × T1; 2 × E1 1 or 2 pairs of copper wire Range: 3700 m without repeater	T1 and E1 services.
SDSL	Digital Subscriber Line	Duplex: 2 × T1; 2 × E1 1 pairs of copper wire Range: to 3000 m	Synchronous services.
ADSL	Digital Subscriber Line	Downstream: 1.5 to 9 Mbit/s Upstream: 16 kbit/s to 640 kbit/s Range: 2700 to 5400 m/1 pair	Internet, VoD, LAN, Video, and multimedia.
VDSL	Digital Subscriber Line	Downstream: 13 to 52 Mbit/s Upstream: 1.6 Mbit/s to 13 Mbit/s Range: 300 to 1350 m/1 pair	Same as ADSL and HDSL.

T1 = 1.544 Mbit/s
E1 = 2.048 Mbit/s
MI = Management Information

The concept of ISDN as conceived in 1976 offers a Basic Rate access of two B channels (2 × 64 kbit/s), one D channel (16 kbit/s) and additional 16 kbit/s for management functions, resulting in a duplex transmission capacity of 160 kbit/s. Data is sent in both directions simultaneously with echo-cancelled transmission. With 2B1Q baseband transmission techniques a bandwidth of 80 kHz is required. The resulting bandwidth efficiency is 2 bit/s per Hz.

Based on ISDN designs during 1992 the first HDSL systems were put into service.

HDSL provides two-way 1.544 or 2.048 Mbit/s transmission over twisted pair subscriber loops up to 3.7 km. Echo-cancelled hybrid duplex 2B1Q transmission is normally used. HDSL systems use two pairs of wires, with each pair conveying half of the payload in both directions (dual-duplex transmission). In addition single-duplex and dual-simplex transmission modes are possible.

SDSL provides the same bandwidth upstream as downstream using one pair. The price paid for maintaining bandwidth symmetry is lower aggregate bandwidth. Data rates from 384 kbit/s up to 2 Mbit/s are possible.

In many applications a higher downstream rate (from the central office end to the remote terminal end) compared to the upstream data rate is required. ADSL is a loop transmission technology that simultaneously transports the following via one pair of wires up to about 5 km:

– Downstream (towards the customer) bit rates up to about 9 Mbit/s;
– Upstream (toward the network) bit rates up to 1 Mbit/s;
– Plain old telephone service.

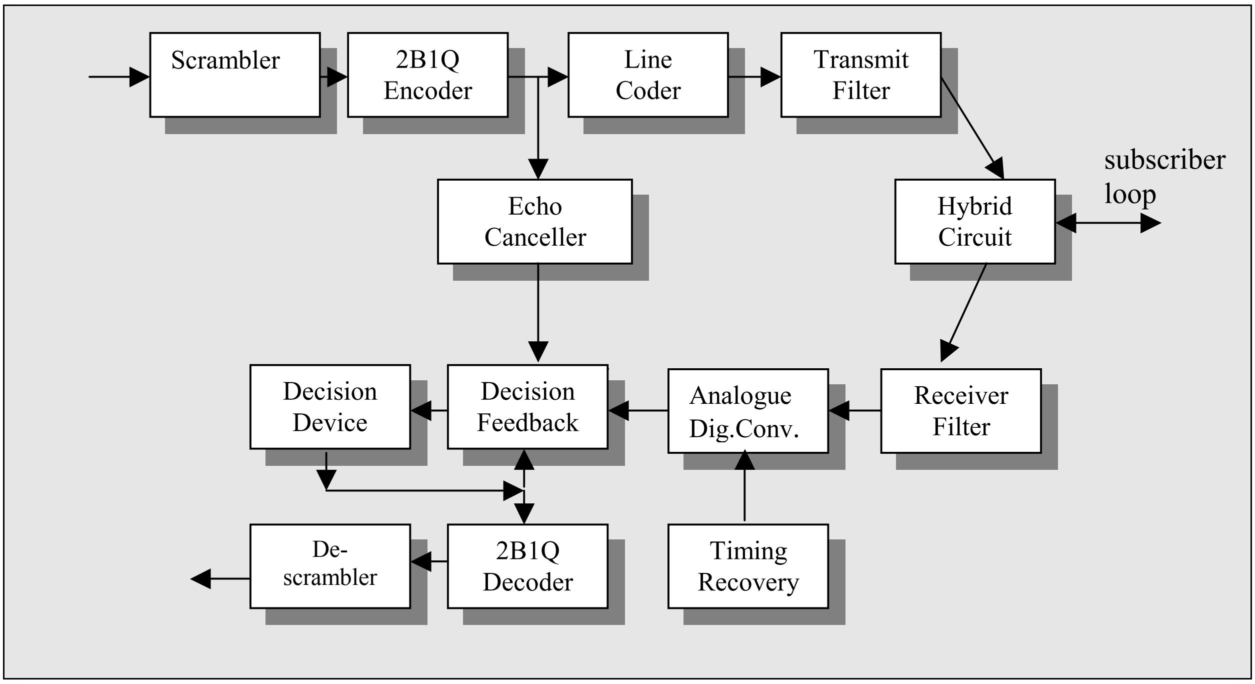
The ADSL system was defined by the ITU-T around 1998, to operate with or without splitter installations. Due to the high bandwidth efficiency of about 15 bit/s per Hz and the simple installation, ADSL systems are expected to be used in large quantities on existing subscriber loops.

VDSL is an extension of xDSL technology to higher bit rates, up to 52 Mbit/s downstream and up to 13 Mbit/s upstream over short distances (the details are still being evaluated within ITU-T SG15).

Figure 6.17 shows the general structure of a 2B1Q DSL transceiver. The incoming data flow from the user terminal is scrambled, to avoid long streams of zeros or ones, every bit pair is encoded into a 2B1Q symbol, thus the symbol rate is half of the bit rate. Two binary signals are converted in one 4-level signal. The symbols are converted to the line code and after filtering sent to the hybrid circuit and subscriber loop.

The incoming line signals are filtered to minimize the out-of-band noise, amplified and regenerated in the analogue-digital converter with the aid of timing signals. The send side 2B1Q symbols are sent to the echo canceller, which models the equivalence of the echo path. Send and receive clocks are synchronized to obtain a stable echo path transfer function. The digitized and echo cancelled received signal is further filtered by the decision feedback circuit and sent to the decision device, which is a four – level threshold detector. After converting to binary signals and descrambling the received information is sent to the user terminal.

Figure 6.17 – General DSL transceiver structure

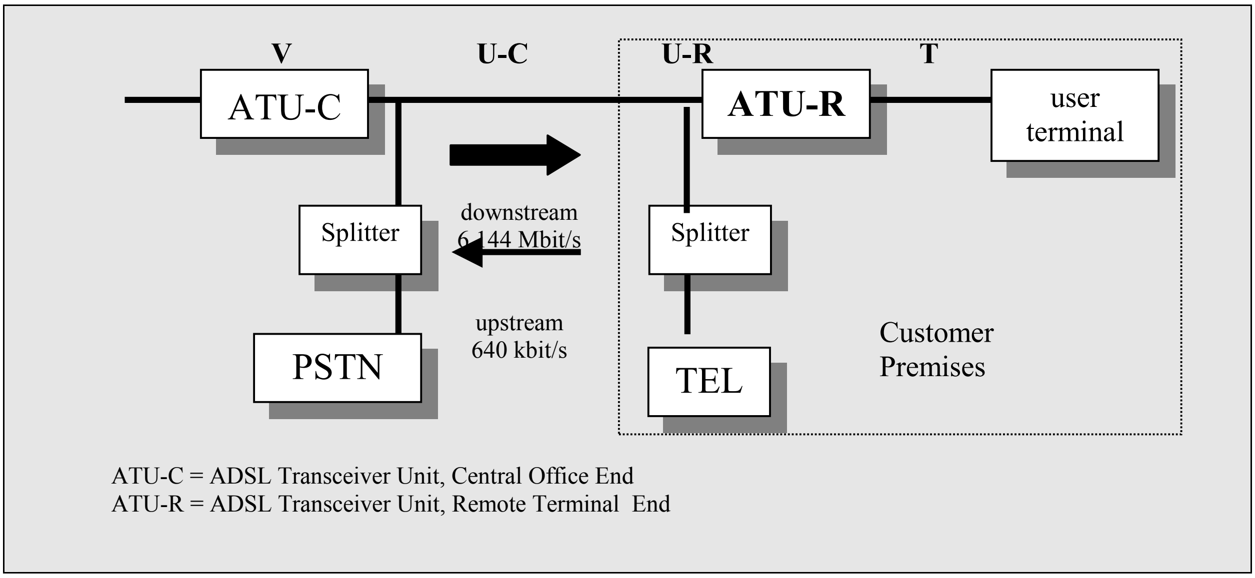


6.4.3 ADSL

Initially, ADSL was specified for Video-on-Demand and ISDN type services. In contrast to other transmission systems with defined bit rates ADSL contains rate adaptation, i.e. the throughput depends on the quality of the transmission media.

Figure 6.18 illustrates the ADSL Reference Model:

Figure 6.18 – ADSL reference model



Two versions of systems are considered as shown in Figure 6.19:

– Full-rate ADSL with a cut-off frequency of 1104 kHz;

– ADSL Lite with a cut-off frequency of 552 kHz.

The available bandwidth of the subscriber loop is divided into frequency bands for:

– analogue POTs or ISDN;

– upstream subcarriers;

– downstream subcarriers.

In addition to the versions shown in Figure 6.19, ADSL Lite and Full Rate ADSL can use echo cancellation, i.e. the frequency band 4-138 kHz is used for both upstream and downstream transmission.

ADSL Lite is expected to replace voice band modems for internet access and will be used in considerable quantities if the following can be achieved:

– Easy end-user installation without splitters and visit by service personnel;

– Long transmission distance;

– Flexible data rates up to 1.5 Mbit/s for users;

– Interoperability and compatibility with Full-rate ADSL.

The requirements for easy installation make splitterless installation an important provision for ADSL Lite. However, in certain cases ADSL Lite demands splitters or additional filters to protect the telephone sets as illustrated in Figure 6.20.

Figure 6.19 – Frequency plan for ADS

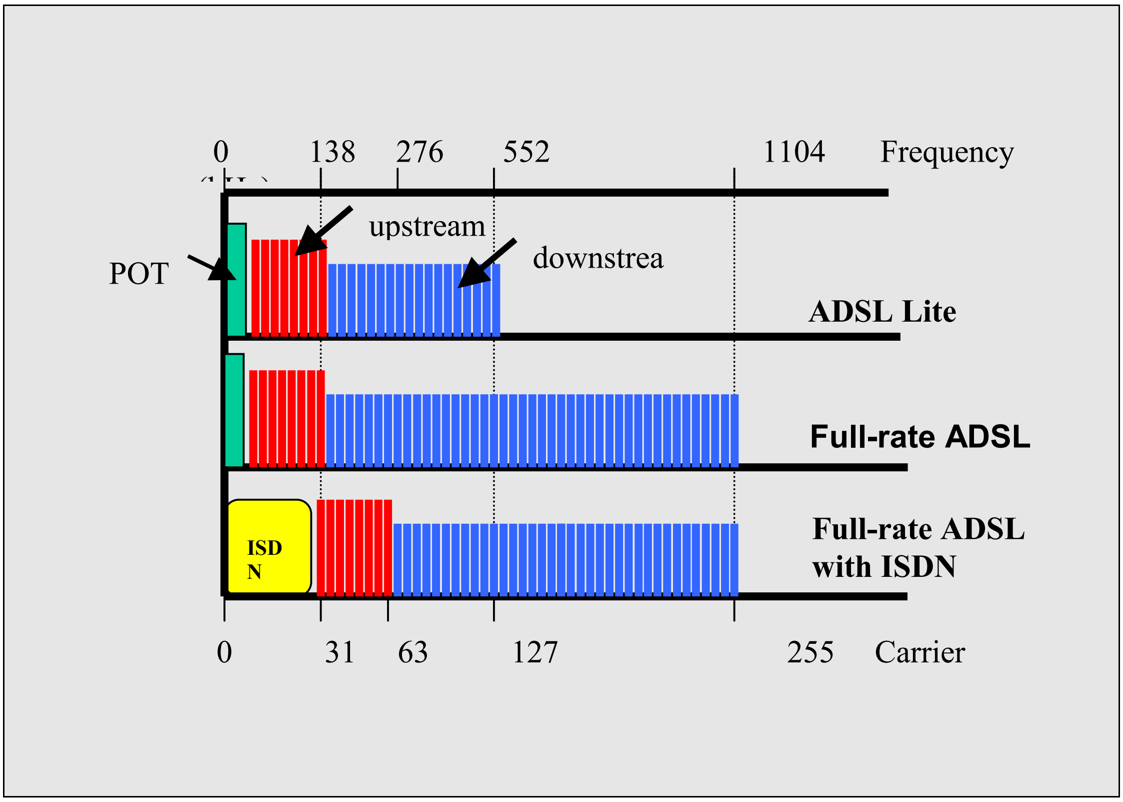


Figure 6.20 – Examples of ADSL lite installation

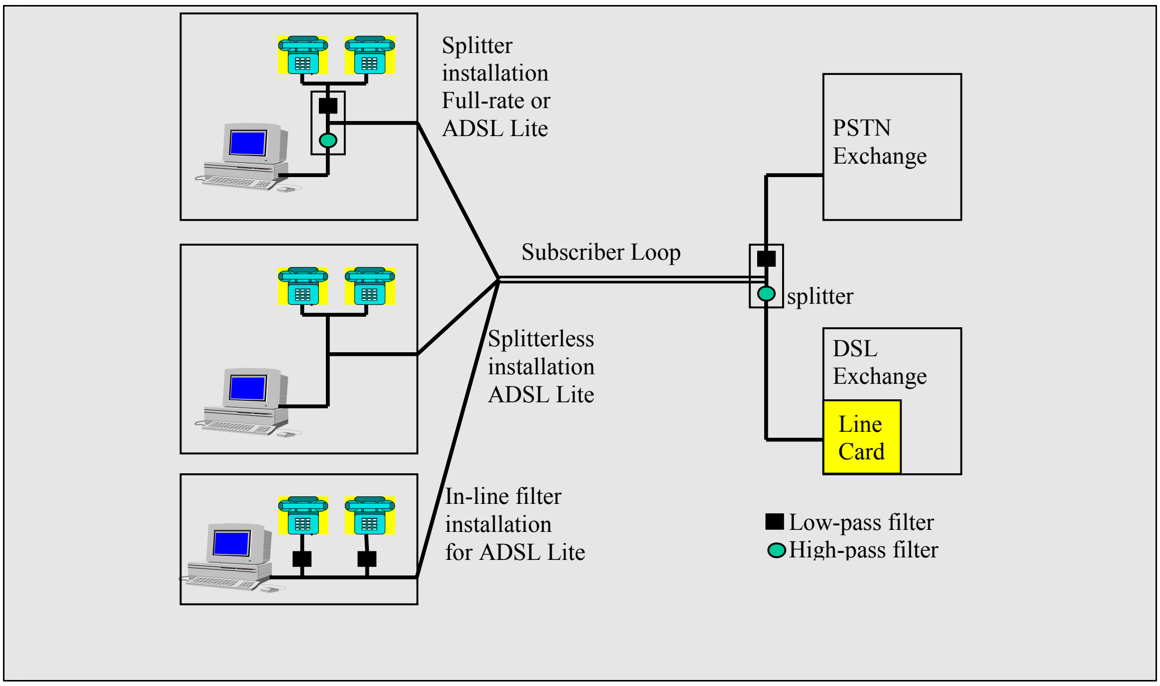


Table 6.6 below illustrates examples of ADSL ranges over loops with typical noise and 0.4 mm wire diameter.

Table 6.6 – ADSL range and rates

0.4 mm wire	Downstream	Upstream	Loop Length
Full-rate ADSL	4 096 kbit/s 2 048 kbit/s 578 kbit/s	120 kbit/s 128 kbit/s 128 kbit/s	2.8 km 3.5 km 4.2 km
ADSL Lite	1 536 kbit/s 1 536 kbit/s	256 kbit/s 96 kbit/s 96 kbit/s	2.8 km 3.5 km 4.2 km

The technology specified for ADSL is based on Discrete Multitone (DMT) transmission.

Figure 6.21 shows a simplified ADSL terminal with the following functions:

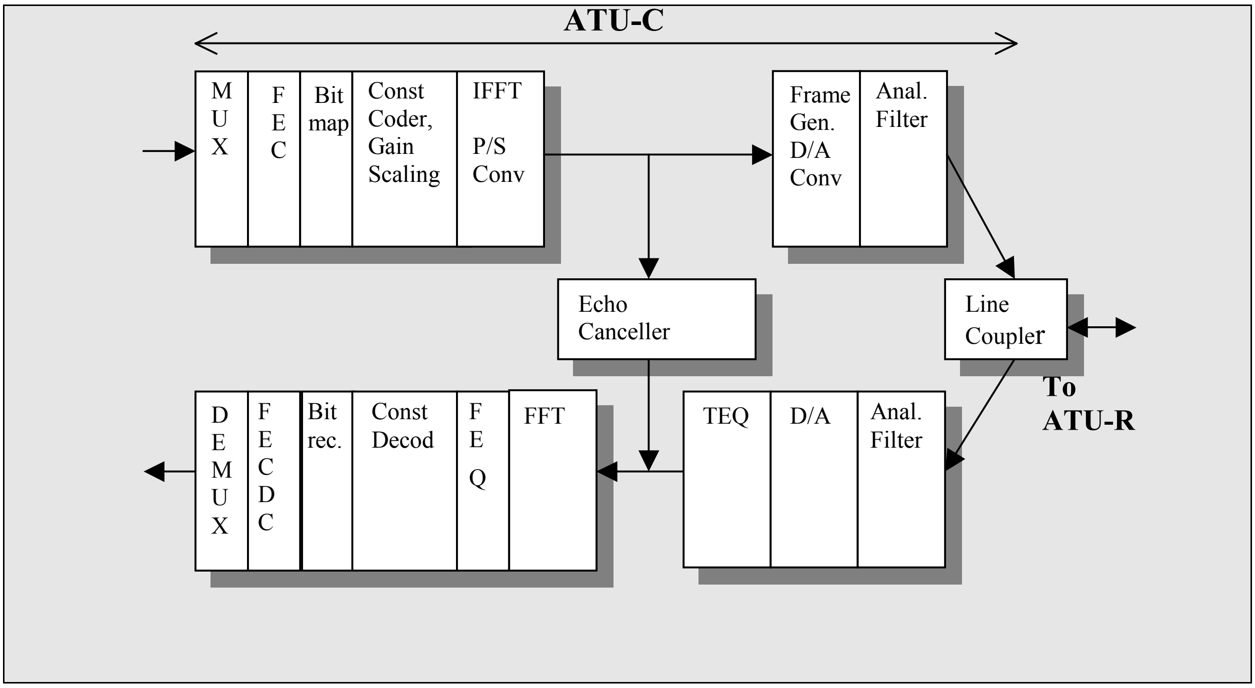
– transmit and receive filtering, automatic gain control, A/D and D/A conversion;

– modulation/demodulation, coding/decoding and bit packing/unpacking;

– Fast Fourier Transform (FFT) and Inverse Fast Fourier Transform (IFFT);

– Adaptive echo cancellation, adaptive channel equalization, symbol/bit conversion and timing recovery.

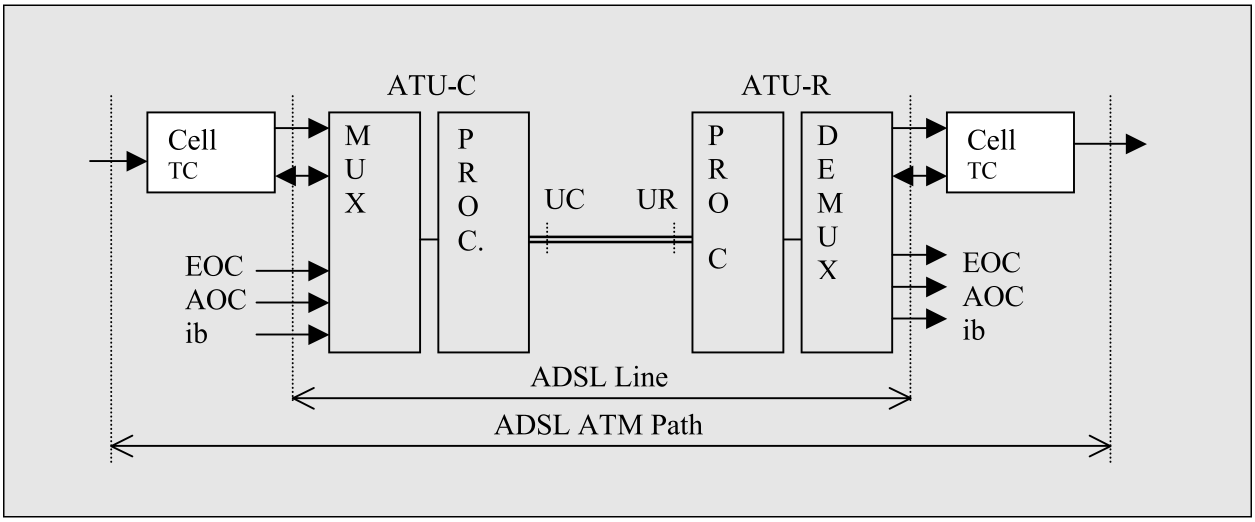
Figure 6.21 – Example of ADSL terminal ATU-C



ADSL systems can be considered as physical layer modulation scheme that provides an asymmetrical physical connection between two endpoints, ATU-C and ATU-R.

Synchronous STM signals and asynchronous ATM signals[1] (or other frame based protocols) can be transmitted across ADSL systems. Figure 6.22 shows the example of an ADSL ATM Path.

Figure 6.22 – ADSL ATM link



The ADSL Link transmits the simplex channel AS = 6144 kbit/s and the duplex channel LSO = 640 kbit/s. In addition for management the Embedded Operation Channel EOC, ADSL Operation Channel AOC and Indicator bits ib, are transmitted.

In order to transport ATM cells a Transmission Convergence Sublayer (TC) has to be provided with the following functions:

– generation and recovery of the transmission frame;

– adaptation of cell flow to physical transmission medium;

– cell delineation and header error correction;

– ensuring that calls are not be dropped during a 3 second interruption.

In addition to ATM cells, frame based protocols, which are widely employed in present networks can be transmitted using ADSL, e.g. the point-to-point protocol (PPP) and the frame-based user network interface (FUNI). STM derived signals can be transmitted across ADSL as multiples of 32 kbit/s.

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1 See more about ATM in Chapter 7 of this Fascicle.

6.4.4 HDSL

The need for HDSL became evident when T1 and E1 transmission systems were used as private lines between Central Office and customer premises. HDSL offers however a number of advantages, such as:

– lower bandwidth requirements;
– lower crosstalk into other systems;
– better diagnostics due to ample overhead, giving reduced maintenance cost;
– operation over 95% of HDSL lines without mid-span repeater;
– typical error rates better than 10^{-9}.

HDSL systems can work in different modes:

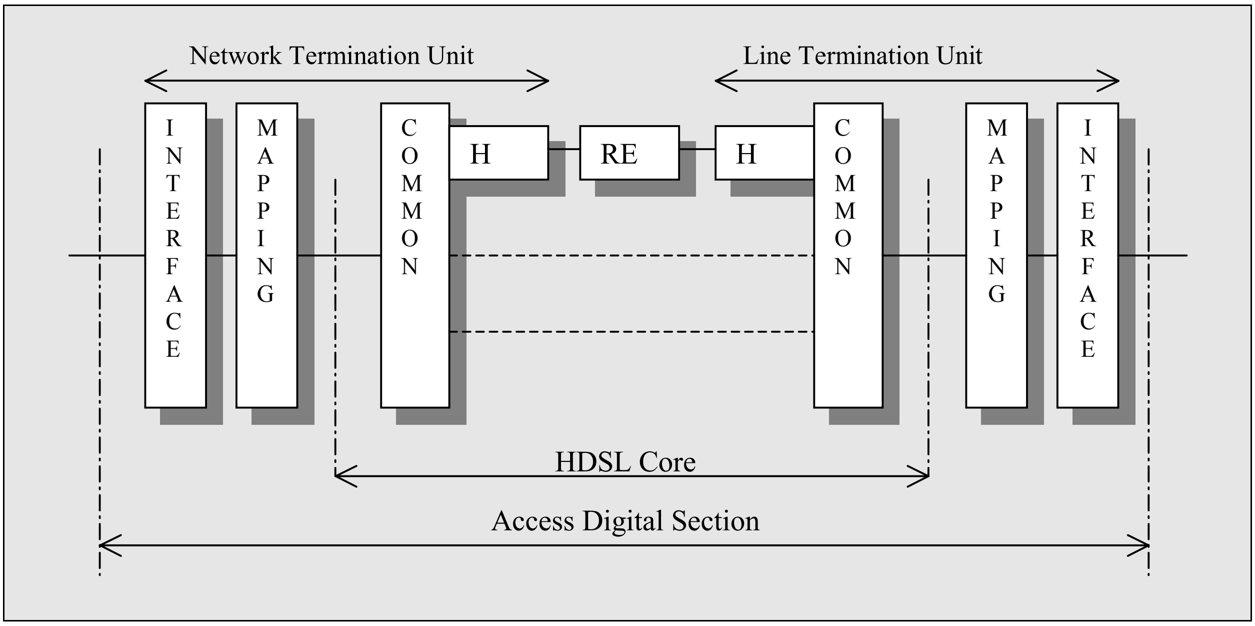
Dual Duplex 1.544 HDSL systems operate on two pairs of wires, each pair conveying 768 kbit/s payload and 16 kbit/s Embedded Operation Channel (EOC) in both directions. 2 048 HDSL systems operate on 2 or 3 pairs. The echo-cancelled hybrid transmission normally uses the 2B1Q line code.

Single Duplex Systems (also called SDSL) operate on one single pair, the two transmission directions are separated either by frequency division multiplexing or echo-cancellation. However due to the wide frequency spectrum required only a limited ranges is possible.

Dual Simplex Systems use two pairs with one pair carrying the full payload in one direction and the other pair the full payload in the opposite direction.

Figure 6.23 shows a simplified HDSL Reference Model. A fully equipped HDSL transceiver transmits over one pair at 2320 kbit/s, over 2 pairs at 1168 kbit/s per pair, or over 2 or 3 pairs at 784 kbit/s per pair. The transceivers are connected via Digital Local Lines (DLL) or connected to Regenerators (REG). The Network Termination Unit (NTU) acts as the master to the slave Line Termination Unit (LTU).

Figure 6.23 – HDSL reference model

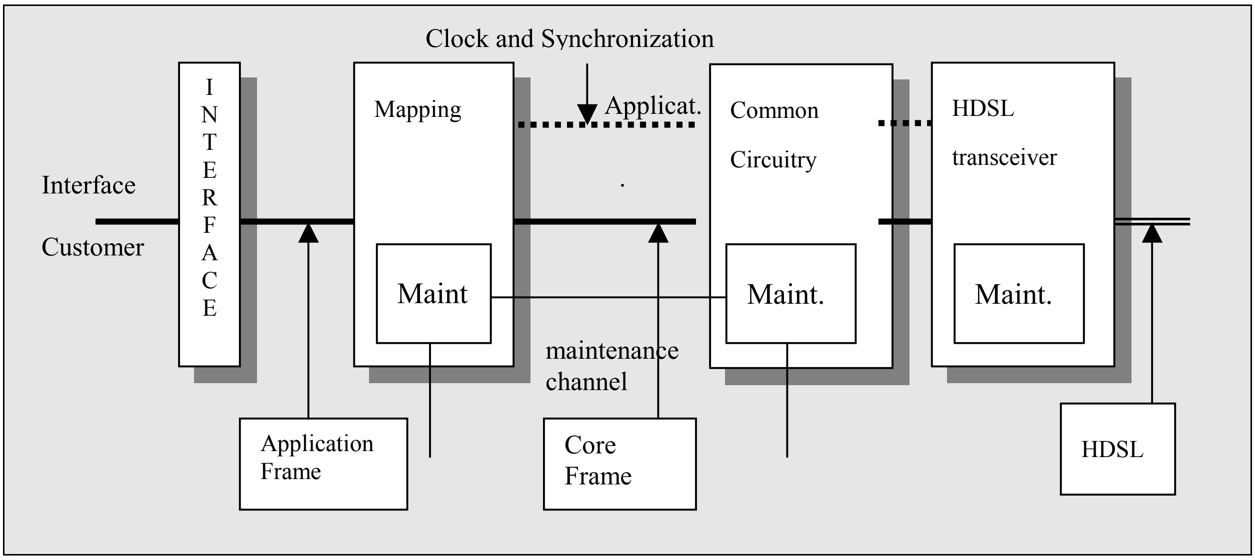


The line code for a Dual Duplex 1544 Mbit/s system is of 2B1Q type leading to spectrum components between 0 and 392 kHz. The dc component of the code can not pass the hybrid transformers which leads to pulse distortion and the subsequent need for equalization.

Figure 6.24 illustrates the simplified function of the HDSL system. The bit stream from the customer side enters the NTU over the Application Interface and is grouped at the Interface in Application Frames. For 2.048 Mbit/s applications the Application Frames are mapped into 144 byte Core Frames. The Core frame information is multiplexed with alignment bits, maintenance and overhead bits in order to obtain HDSL Frames which are then transmitted via Digital Local Lines using 2B1Q line codes.

At the receiving side, data within the received HDSL frames is demultiplexed to obtain again the Core Frames. Core Frames are passed to the mapping function to restore the Application Frame and transmitted to the Application Interface at the network side.

Figure 6.24 – HDSL transceiver function



6.4.5 VDSL

Very High Bit Rate Subscriber Line, VDSL, is an extension of xDSL technologies to higher bit rates. VDSL can support voice, data and video simultaneously in addition to future applications such as high definition Television and high-performance computing. VDSL downstream rates are up to 52 Mbit/s and upstream data rates up to 13 Mbit/s. Such high data rates can only be transmitted over short distances. It is anticipated that the evolution of the existing telephone line plant will bring more fibre links to replace the physical subscriber loops and that the remaining subscriber loops will be in the range of 1 km. This distance is suitable for VDSL systems. Figure 6.25 shows the reference model of a VDSL-based broadband transmission system.

VDSL systems are normally connected to optical access networks, e.g. Fibre to the Curb installations. VDSL loops are terminated in VDSL Terminal Units, VTU-C at the network side and VTU-R at the remote user side. VTU-C can be located in the Optical Network Unit (ONU) which terminates the Optical Distribution Network ODN. With service splitters PSTN or ISDN signals can be separated from the VDSL signals.

The information transfer is organized in VDSL frames containing payload and separate upstream and downstream error control mechanisms. The line codes can be QAM (8 bits per symbol) or CAP leading to the parameters shown in Table 6.7 below:

Figure 6.25 – VDSL reference model

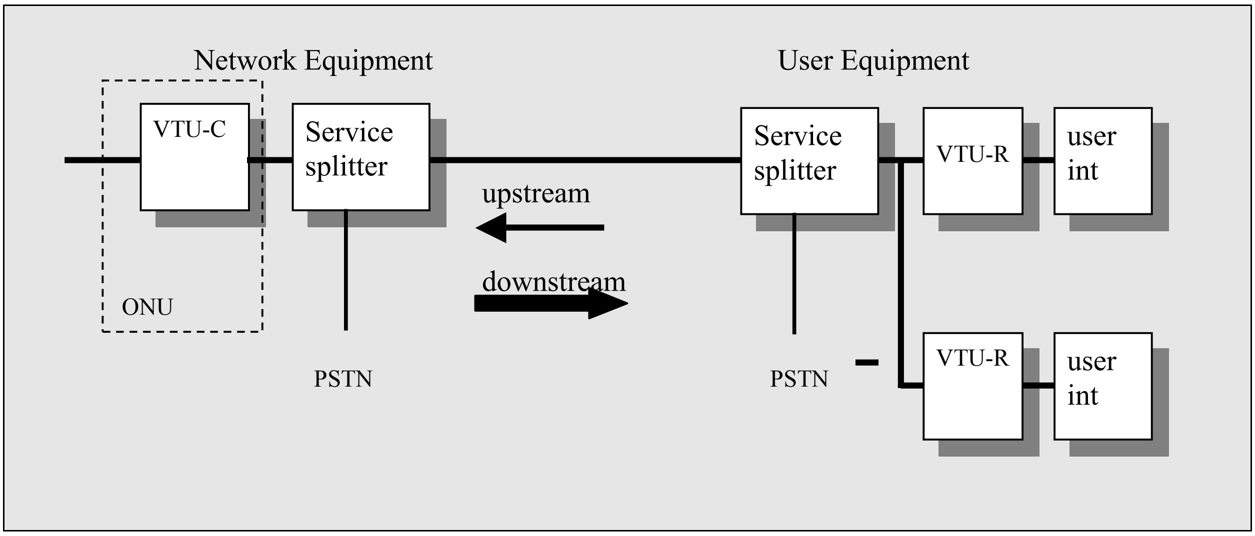


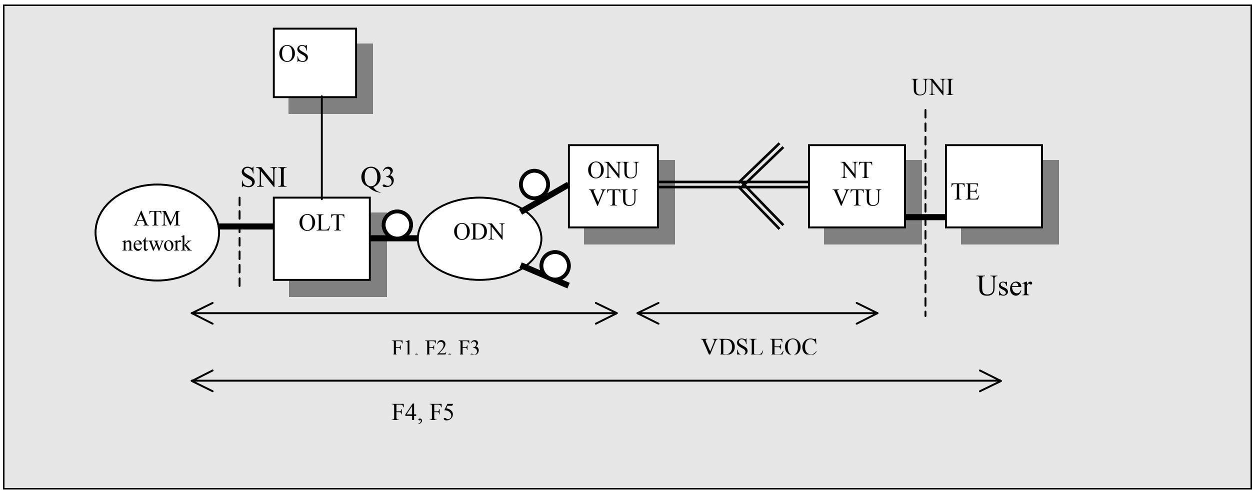
Table 6.7 – Typical VDSL data rates

Loop length km	Downstream rate Mbit/s	Frequency spectrum MHz
1.5	13	2-5.8
1.0	26	2-9.8
0.3	52	2-17.5
1.5	1.6	1.2-1.4
1.0	3.2	1.1-1.6
0.3	6.5	0.8-1.8

The frequency bands for short wave and amplitude modulated medium wave radio stations, and the public safety/distress bands are in the same range as the bandwidth occupied by VDSL signals, which can lead to signal interference. The VDSL throughput is mainly limited by background and self-NEXT noise in the loop plant.

Figure 6.26 shows the example of a combined optical and VDSL network. The Optical Distribution Network ODN is located between the Optical Line Termination OLT and the Optical Network Unit ONU, which also contains the VTU-C. The VDSL link is terminated in the Network Termination NT, which also contains the VTU-R. For management purposes, an Operation System OS is connected via the Q3 management interface to the OLT. The management of the ATM links is divided between the ATM layer (F4 and F5) and the Physical layer (F1, F2, F3). Management information related to the VDSL link (contained in VDSL Embedded Operation Channel) is transported to the OS via ATM.

Figure 6.26 – Example of VDSL ATM network management



6.4.6 Network Examples

Since xDSL is a transmission technology primarily at the physical layer it can support a variety of network protocols, services and applications, mostly in the access part of telecommunication networks. At this stage there is little consensus regarding which network architecture is most suitable to support xDSL as an access technology. For the time being ADSL is considered to satisfy the majority of applications, as 6 to 8 Mbit/s can be transmitted to customer installations on most existing subscriber loops.

Figure 6.27 illustrates an example the transport of Internet Protocol (IP) traffic.

According to specifications of the ADSL Forum, IP packets can be transported directly over ADSL links without the need for additional protocols. In addition to the corporate application shown in the Figure, IP can be used for residential applications with Personal Computers (PC) connected to the ADSL remote side. Access to the IP network can be established via a Point-to-Point Protocol (PPP) over an ADSL connection.

Figure 6.27 – ADSL IP transmission

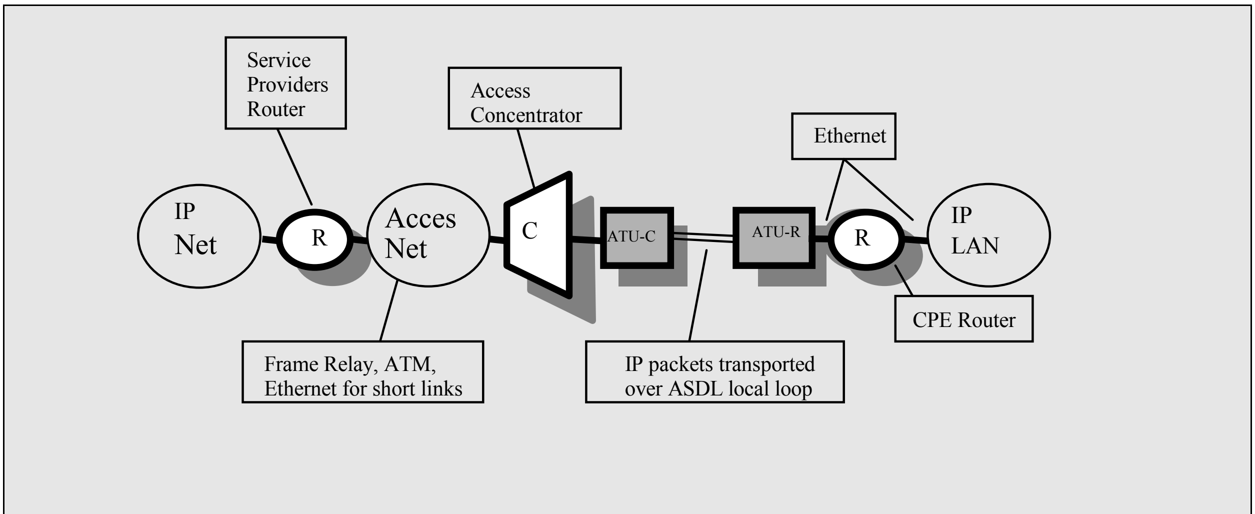


Figure 6.28 illustrates the ATM transmission via ADSL. The ADSL Forum has specified how ATM cells can be transmitted over ADSL, from the customer terminal to the ATM network. However, ATM applications operating at data rates below 25 Mbit/s via multiple ADSL links results in a rather extensive overhead. In the case of 155 Mbit/s, connection over fibre is preferable.

Figure 6.28 – ADSL/ATM transmission

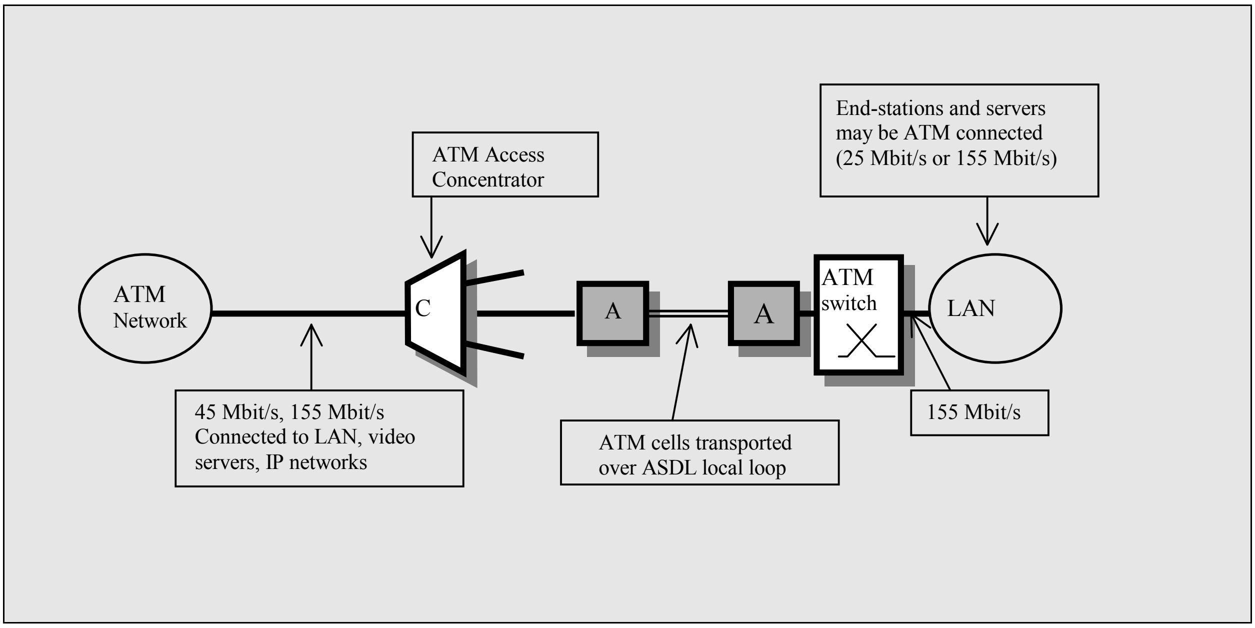


Figure 6.29 shows the transmission of synchronous traffic via ADSL links. Two or three TU-12 signals can be transmitted from the ATU-C to ATU-R (each TU-12 signal corresponds to 2.304 Mbit/s)

Figure 6.29 – ADSL synchronous transmission

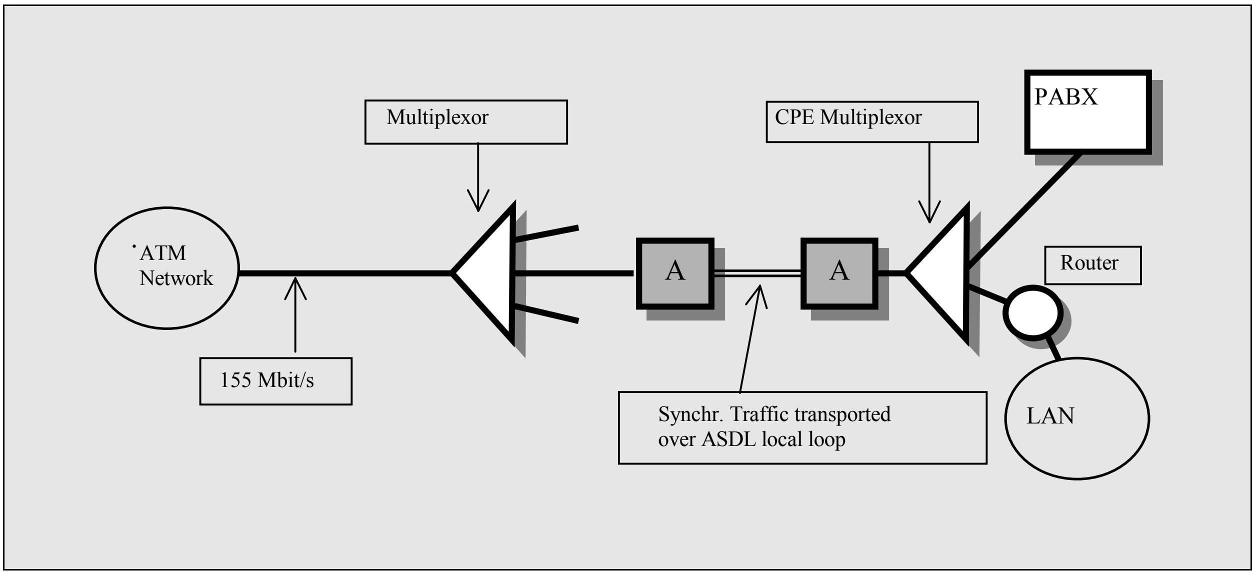
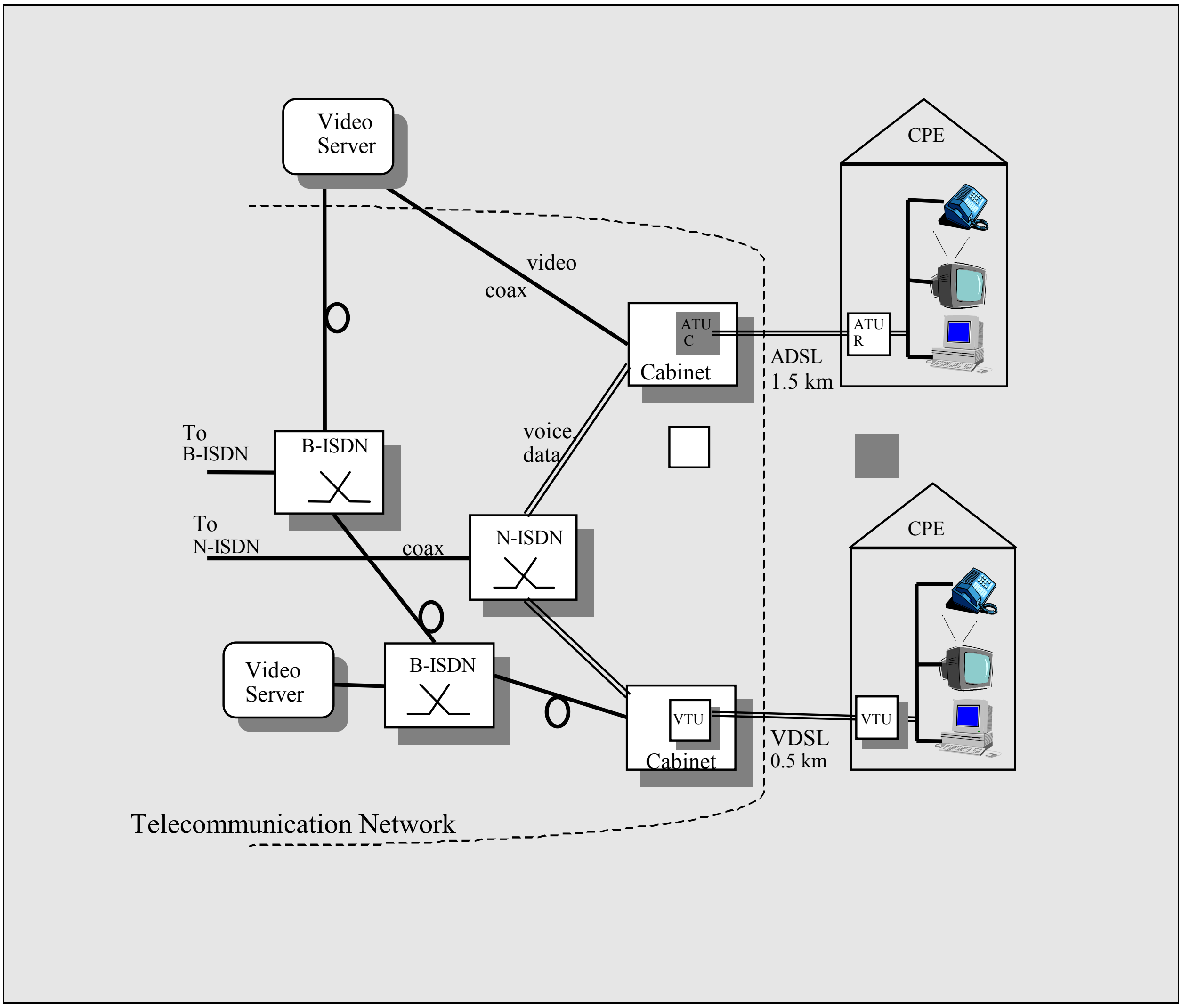


Figure 6.30 shows the connection of Customer Premises Equipment (CPE) to a telecommunication network with ADSL and VDSL links. From an external video server digitalized video signals reach:

- Cabinet 1 via a coaxial cable. The CPE is connected via an ASDL link;
- Cabinet 2 over fibre cable and B-ISDN switches. The CPE is connected via a VDSL link.

Data and voice signals are transported over the N-ISDN network to a N-ISDN switch. The signals reach Cabinets 1 and 2 over normal cable. In addition an internal video server can be connected to Cabinet 2 via a B-ISDN switch.

Figure 6.30 – ADSL/VDSL network example



6.4.7 ITU-T Recommendations

The study of xDSL technology started in 1993 with ANSI standardization work, defining Discrete Multitone (DTM) for ADSL operation. Cooperation with ETSI took care of specific European requirements.

In 1997 the ITU-T began defining a series of Recommendations for DSL systems and at present (year 2000) Recommendations have been elaborated by Study Group 15 dealing with functions, management, handshaking and test principles for ADSL, HDSL and VDSL systems. A study of the xDSL related Recommendations of the G.99x Series reveals the complexity of the DSL systems, nearly 1000 pages are required to define the systems! The family of G.99x Recommendations include:

G.991.1 High bit-rate Digital Subscriber Line (HDSL) Transceivers.

G.992.1 Asymmetrical Digital Subscriber Line (ADSL) Transceivers.

G.992.2 Splitterless Asymmetrical Digital Subscriber Line (ADSL) Transceivers.

G.994.1 Handshake Procedures for Digital Subscriber Line (DSL).

G.995.1 Overview of Digital Subscriber Line (DSL) Recommendations.

G.996.1 Test Procedures for Digital Subscriber Line (DSL) Transceivers.

G.997.1 Physical Layer Management for Digital Subscriber Line (DSL) Transceivers.

In addition to these Recommendations Study Group 13, under the concept of the Global Information Infrastructure (GII), has developed several general Recommendations, e.g. Rec. Y.120 containing scenarios which include xDSL technology.

6.4.8 Outlook

ADSL is regarded as the basic DSL service option for the residential market. As an ADSL system offers fast downstream and slower upstream data rates, it is ideal for Internet access. Competitive Local Exchange Carriers (CLECs) consider the availability of ADSL as an ideal opportunity to get into the data market quickly and compete with the incumbent carriers. To deploy ADSL, however, Line cards or Digital Subscriber Line Access Multiplexors (DSLAM) would have to be installed near to central office belonging to the incumbent carriers. In addition, because of the quality of home networks and in-premises wiring, which impacts the maximum data rate and range of the transmission system, many situations are not appropriate to splitterless technologies.

HDSL allows the provision of high-speed digital services over existing copper subscriber loops over 2 to 4 parallel pairs. For distances up to about 3 km HDSL is a quick and cost-effective option for the deployment of duplex T1 or E1 systems. Extensive work is in progress to define a HDSL system using only one pair, called “Next Generation HDSL” HDSL2, or SHDSL. To achieve this goal the system performance and gain have to be increased and the effects of NEXT have to be mitigated with spectrum shaping.

The investigations compare different line codes, filters and resulting transceiver complexity. HDSL2 may be about 8 times more complex than HDSL!

ADSL is likely to be an interim step towards VDSL, which will provide the extra bandwidth needed for full broadband services. The success of VDSL will depend on fibre being extended much closer to the customer, e.g. upgrading of feeder routes to fibre.

6.4.9 Literature

[1] *Reference book on xDSL*, ITU-D Study Group 2, Question 12/2, under preparation, 2000

[2] Albin Johansson, ADSL Lite, *The broadband enabler for the mass market*, Ericsson Review No. 4, 1998.

[3] Thomas Starr *Understanding Digital Subscriber Line Technology*, Communication Engineering and Emering Technologies Prentice Hall PTR, NJ 07458, 1999

[4] Dr. Walter Y. Chen *DSL, Simulation Techniques and Standards*, Macmillan Technical Publishing, Indianapolis, Indiana, 1998.

[5] Prof. Dr. Z. Petrovic *Digitalne pretplatnic ke petlje*, Telfor International Conference Belgrade 1999.

6.5 Abbreviations

2B1Q	2 Binary/1 Quaternary (line coding technique)
AAL	ATM Adaptation Layer
ABR	Available Bit Rate Service
A/D	Analogue/Digital (conversion)
ADSL	Asymmetric Digital Subscriber Line
AOC	ADSL Operation Channel
AOM	Administration, Operation, Maintenance
ASM	ATM Service Multiplexer
ATM	Asynchronous Transfer Mode
ATU-C	ADSL Transmission Unit – Central office
ATU-R	ADSL Transmission Unit – Remote
AU	Administrative Unit
AUG	Administrative Unit Group
B-ISDN	Broadband-Integrated Services Digital Network
BB	Baseband modem
C	Container
CBR	Constant Bit Rate Service
CPE	Customer Premises Equipment
D/A	Digital/Analogue (conversion)
DLL	Digital Local Line
DMT	Discrete Multitone

DSL	Digital Subscriber Line
DXC	Digital Cross Connect
DWDM	Dense Wavelength Division Multiplex
EOC	Embedded Operation Channel
FDM	Frequency Division Multiplex
FEC	Forward Error Correction
FEXT	Far End Crosstalk
FFT	Fast Fourier Transform
FUNI	Frame-based User Network Interface
HDSL	High speed Digital Subscriber Line
HVC	High Order Virtual Container
IFFT	Inverse Fast Fourier Transform
ISDN	Integrated Services Digital Network
LAD	LAN Access Device
LAN	Local Area Network
LVC	Low Order Virtual Container
MI	Management Information
MSSP	Multiplex Section Shared Protection
NEXT	Near End Crosstalk
N-ISDN	Narrowband-Integrated Services Digital Network
NNI	Network Node Interface
NT	Network Termination
NTU	Network termination Unit
OADM	Optical Add-Drop Multiplexer/Demultiplexer
ODN	Optical Distribution Network
OH	Overhead
OLA	Optical Line Amplifier
OLT	Optical Line Termination
OS	Operation System
OSC	Optical Supervisory Channel
PC	Personal Computer
PCM	Pulse Code Modulation

PDH	Plesiochronous Digital Hierarchy
PHY	Physical Layer
POTS	Plain Old Telephone Service
PPP	Point-to-Point Protocol
QAM	Quadrature Amplitude Modulation
RTP	Receive Transponder
SDH	Synchronous Digital Hierarchy
SDSL	Digital Subscriber Line
SONET	Synchronous Optical Network
STM	Synchronous Transport Module
TC	Transmission Convergence sublayer
TE	Terminal Equipment
TMN	Telecommunications Management Network
TP	Send Transponder
TU	Tributary Unit
TUG	Tributary Unit Group
UBR	Unspecified Bit Rate Service
UNI	User Network Interface
VBR	Variable Bit Rate Service
VC	Virtual Container (SDH)
VC	Virtual Channel (ATM)
VCI	Virtual Channel Identifier
VDSL	Very high speed Digital Subscriber Line
VP	Virtual Path
VPI	Virtual Path Identifier
VTU	VDSL Terminal Unit
VTU-C	VDSL Transmission Unit – Central office
VTU-R	VDSL Transmission Unit – Remote
xDSL	generic term for all types of DSL equipment

CHAPTER 7

7 ATM technology

7.1 Introduction

It is believed that ATM is the culmination of all developments in switching and transmission in the last ten years. Communications evolved in history towards the concept of the asynchronous transfer mode. The motivations were very strong and, in order to understand them, a brief inspect of communications evolution is necessary.

Communications started in the nineteenth century with the invention of the telephone, capable of transmitting analogue voice signals. The users were connected via switches across the network to form a circuit. This type of switching is known as *circuit switching* and it provides a permanent allocation of channels or bandwidth between the connection end points. In analogue telephony the only way of sharing the common media was based on *frequency division multiplex,* which appeared in 1925. The first coaxial cables, installed in 1936, provided wide bandwidth and better performance concerning the signal to noise ratio.

Digital communications, which came in the late 1960's, was introduced simultaneously with the *time division multiplexing*. Soon after that a problem with the interconnection of computer systems over the communication networks arose. Modems, generating the analogue signals compatible with the public switched telephone network, were used. Data are transferred in bursts separated by silence intervals. This is why the constant connection provided by the circuit switching was not the optimal one. The solution to the problem was to split the data into discrete units, packets, and to send them individually across a network. Packets contain a considerable overhead involved in error recovery, redundancy enhancement and routing information. So *packet switching* has been developed.

Time division multiplexing, if used in switching, gives an opportunity for *multi-rate switching* where a station attaches to the network by a single physical link which carries multiple fixed data rate channels (B-channels – 64 kbit/s). Traffic on each channel can be switched independently through the network. This principle has also been used in *Integrated Services Digital Network* (ISDN). Though it offered a number of data rate choices, these are fixed and thus, not very efficient for variable bit rate (VBR) transmission.

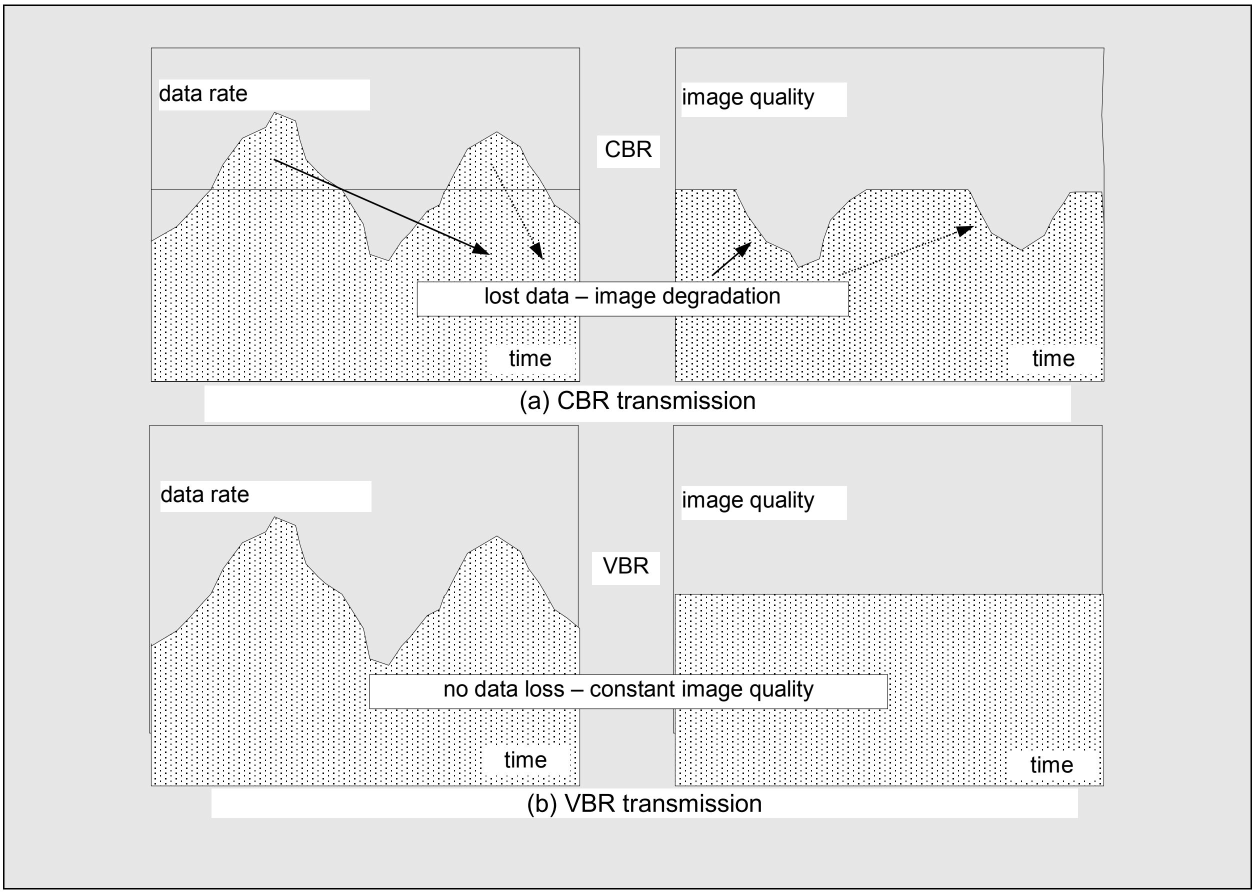
The same byte organization, which enables multirate switching, is present in the *synchronous digital hierarchy network* (SDH). It should be noted that SDH uses synchronous time division multiplexing. It provides means for controlling the network resources in order to deliver a guaranteed quality of service (QoS), but at the expense of inefficient use of those resources.

In order to accommodate different data rates in modern high speed communication networks, *frame relay* appeared as a promising solution. It is essentially identical to packet switching with the exception of the variable length packets, and is designed to operate at up to 2 Mbit/s. Packets in the frame relay systems include less overhead than in the previous systems. In the context of supporting VBR traffic, it has improved performance.

Real-time applications such as voice or video tend not to be very bursty, producing the constant bit rate at the output. It is efficient to assign a fixed bandwidth to a single source. Available capacity is shared between such sources on an equal-portion basis.

The tremendous need for delivering more data, especially for video applications resulted in various compression techniques. Consequently, the bit streams produced in many new applications are variable. By assuring variable bit rate (VBR) transmission, satisfactory quality of service can be obtained. For instance, less significant details in image (or video) transmission are compressed and the bandwidth saved for more important details. In case of VBR transmission, constant quality instead of constant rate may be produced, Figure 7.1.

Figure 7.1 – Constant rate vs. constant quality, for
(a) CBR transmission
(b) VBR transmission



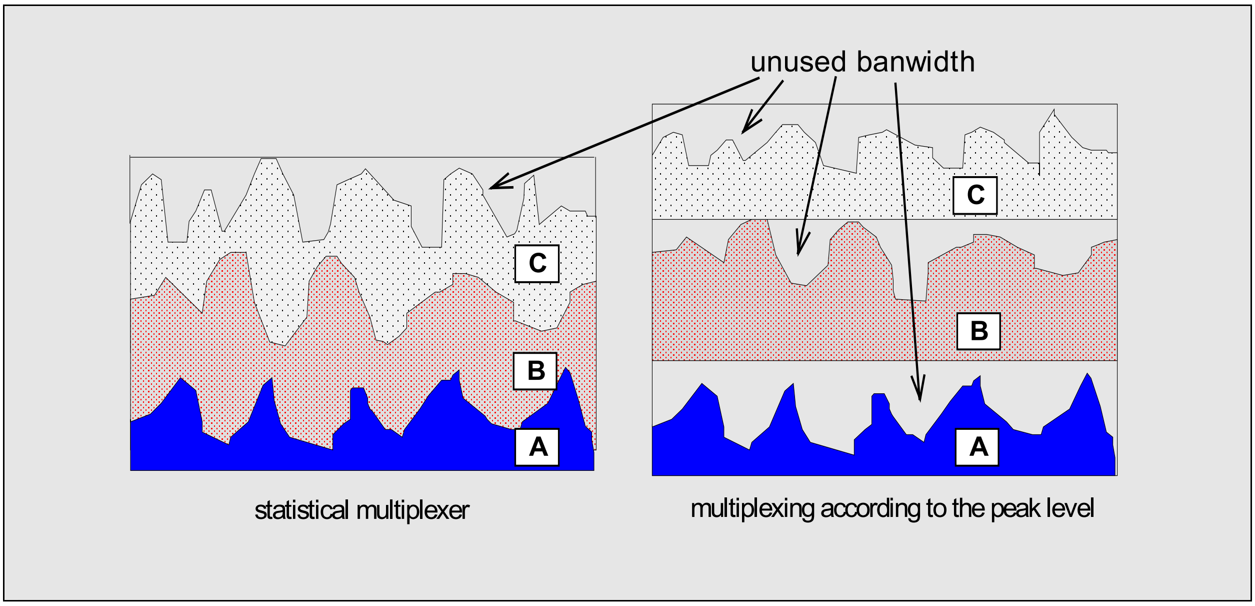
The introduction of new services at the end of 1980 required a new technology which could support multiple services, deliver the necessary high-speed transmission rates, allow for bandwidth on-demand and offer end-to-end control for efficient management. So, traditional networks had to be specialized in order to meet these requirements. Various network types have been developed introducing specific interfaces, facilities and support requirements. With the development of optical fibre technology, a high speed transmission media with low susceptibility to noise was obtained. It enabled transmission of different types of broadband services consuming significant bandwidth. This was the initial power for the *Broadband ISDN* (B-ISDN) foundation.

Frame relay evolved to cell relay, using fixed sized packets of short length, cells. Having a low noise high speed transmission media, and higher layer network control, it provides minimum overhead for error control. Cell relay allows for the definition of virtual channels with data rates dynamically defined. So, cell relay can be viewed as a progression from: circuit switching for constant bit rates, and packet switching for the variable bit rates.

To ensure that information is not lost, any signal should be transferred at the peak natural rate. As a consequence, at times of lower information rate (for example in which redundancy was present in the original signal and thus coding techniques have saved bandwidth) network resources are not be used efficiently. To resolve this, multiplexing of different signals has been proposed.

ATM provides an efficiency gain through statistical multiplexing of dynamic user traffic. It allows multiple connections to share the output-port bandwidth, producing a high utilization of the available resources, see Figure 7.2. Having inputs with fluctuating bandwidth requirements that merge at one output queue, a statistical multiplexer output cell rate is less than the sum of the peak input cell rates. To avoid cell loss it is necessary to use a buffer for storing the excess queued cells. It was believed that appropriate buffering should decrease the burstiness (peak to average cell ratio), smoothing the aggregate cell stream (output of a statistical multiplexer). However, it has been shown that the aggregate stream is as bursty, as the inputs are. As a result, achieving a statistical multiplexing gain requires a much larger buffer at every contention point. Buffer dimensioning is the subject of a low delay-low cell error ratio compromise.

Figure 7.2 – Statistical multiplexer vs. peak level based multiplexer



According to ITU-T Recommendation I-120, “The main feature of the ISDN concept” and thus the B-ISDN too “is the support of a wide range of voice and non-voice applications in the same network. A key element of service integration … is the provision of a range of services using a limited set of connection types and multipurpose user-network interface arrangements.”

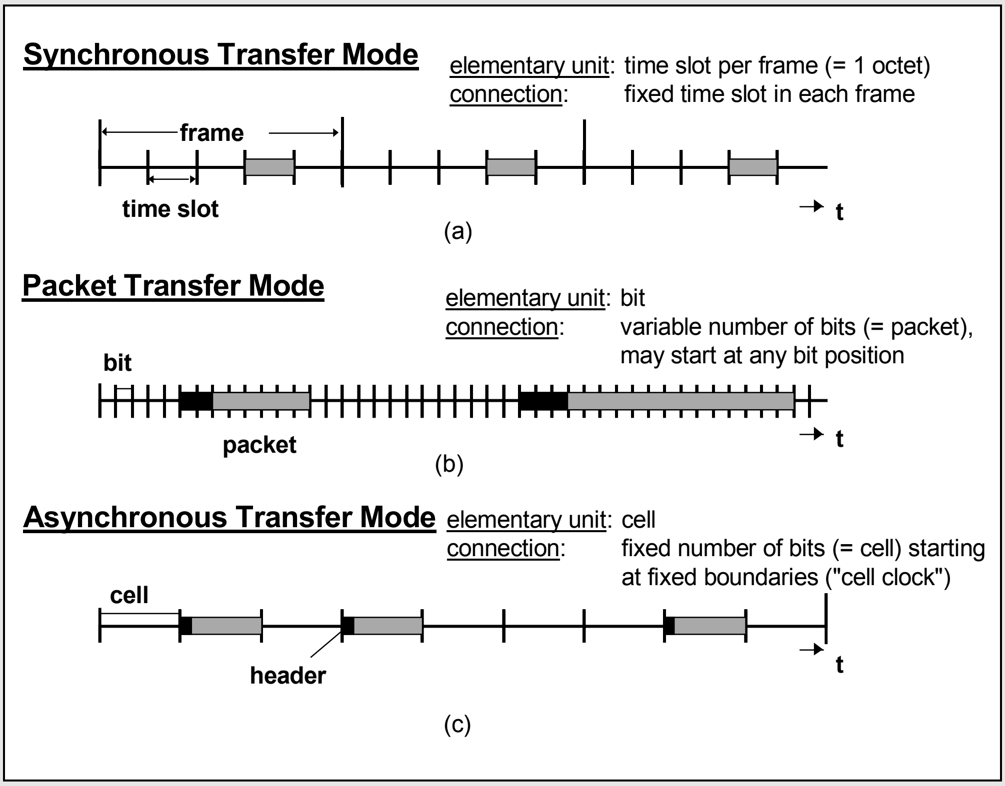
The ATM is a transfer mode for implementing the B-ISDN. By the term ‘transfer’ both the transmission and the switching of the information (with reasonable delay and complexity) are included. In order to explain the ATM properties, three different types of transfer mode are presented in Figure 7.3.

Today, the standard switching technique for narrow band signals (which are mainly voice connections) is by the switching of synchronous channels for the duration of the connection. The switching is realized by combinations of space and time switching stages in exchanges. In *synchronous transfer mode* (STM) each connection is periodically presented as a fixed-length word (also named "time slot"), Figure 7.3 (a).

Data applications use packets (information blocks). In a *packet transfer mode* (ITU-T, I.113), the transmission and switching functions are achieved by packet-oriented techniques (by their address only, without any relation to time), so as to dynamically share resources among different connections, Figure 7.3 (b).

In the *asynchronous transfer mode* all information to be transferred is packetised into fixed-size slots called **cells,** and it operates in connection-oriented mode. These cells have a 48-octet information field and a 5 octet header containing routing and control information. The transfer is asynchronous in the sense that the recurrence of cells containing the information from an individual user for a single service is not necessarily periodic, Figure 7.3 (c).

Figure 7.3 – Comparison between three various transfer modes



The ATM technology is targeted at eliminating the duplication of hardware and software requirements. Thus, a single network should provide higher link efficiency, simplified operations, maintenance, services provisioning, reduced equipment costs and a flexible allocation of network resources.

The ATM uses the Asynchronous Time Division (ATD) principle in which a transmission capability is organized in non-dedicated slots filled with labelled cells with respect to instantaneous need of each application. Each application source, as it may have a variable bit rate, defines its own transmission rate.

ATM complies with three basic requirements for future services:

Future services require high transmission rates, reaching more than 100 Mbit/s. They will be used for fast document transmission, fast processor connections or video transmissions.

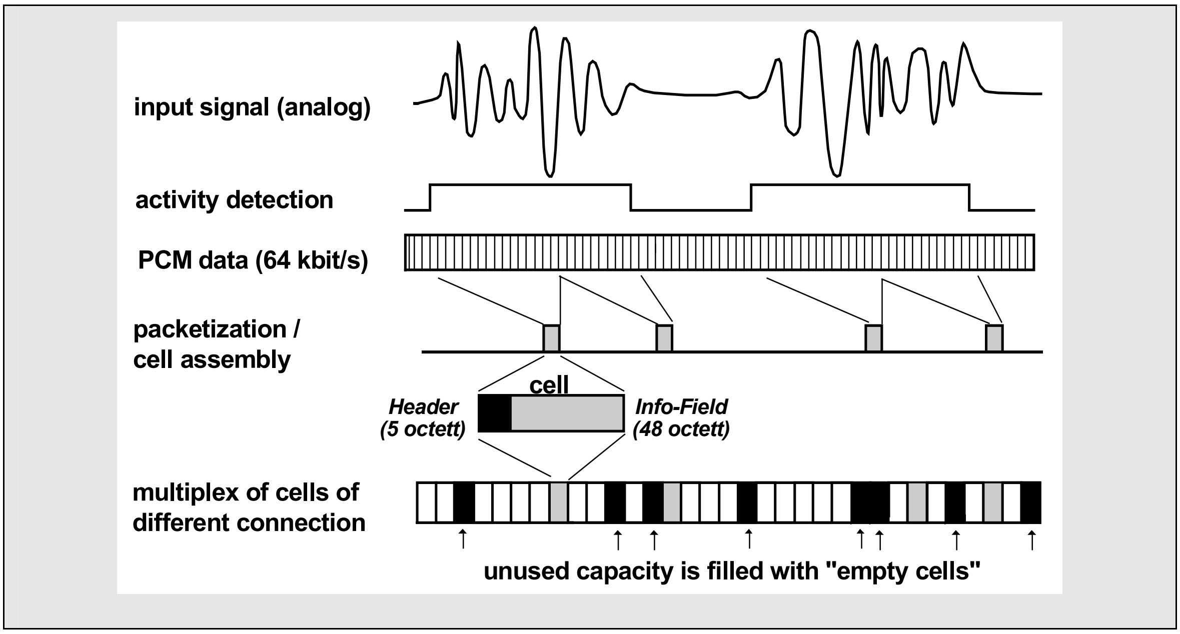
Many services need a variable transport capacity that can be defined for each connection individually. Depending on the traffic characteristic of the service, both continuous and packetised information may have to be transmitted.

The third requirement follows the need for variable bit rates during the connection. Interactive services have phases with very high bit rates during the information transfer and phases with nearly no information flow during the information processing or view (bursty traffic). Variable bit rate coding generates different bit rates during the connection.

ATM cells can be transported on different transmission systems. The only requirement is that bit sequence independence is guaranteed, meaning that there are no restrictions on allowed cell information. The ITU-T defined two options for the user-network interface, one based on the SDH, and the other on pure cell multiplexing.

The source signals are packetised in the terminal, or for conventional terminals in a separate terminal adapter, into ATM-cells. In ATM-systems (exchanges, multiplexers, concentrators) cells of different connections are statistically multiplexed. Unused transmission capacity will be filled with empty cells. Figure 7.4 shows an example of the process how analogue signals are packed into cells and these cells are multiplexed with cells of other connections.

Figure 7.4 – Packetization/cell assembly process.



In an ATM network, new effects show up which are not known in synchronous networks. For example it takes 6 ms to fill one cell of a 64 kbit/s data stream. In the network, these cells are multiplexed with other cells and modified in the switching node. So, additional, rather than fixed delays appear. For services with constant bit streams, appropriate measures have to be provided.

Furthermore, cells can be lost because of bit error, buffer overflow or by the action of a policing function which supervises the cell stream for compliance with agreed parameters. A lost cell represents the loss of 48 bytes of information. The source coding has to cope with this kind of error.

7.2 Virtual connections – Virtual channel and virtual path

The ATM is a ***connection-oriented*** technique with virtual connections between the termination points. Basic terms concerning virtual connections are defined in ITU-T Recommendation I.311.

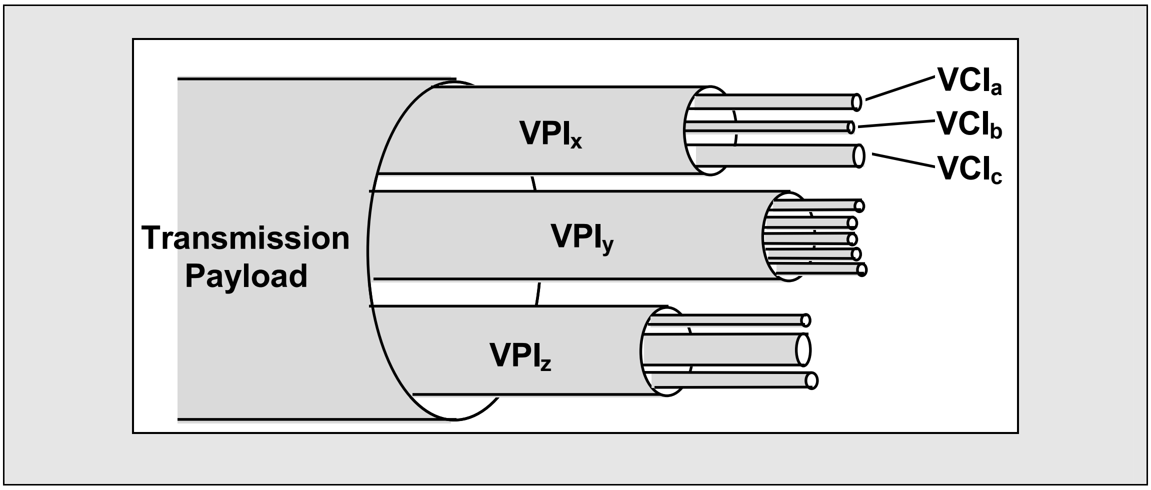
Virtual Channel

The virtual channel (VC) is a generic term used to describe a unidirectional communication capability for the transport of ATM cells. Any virtual channel obtains its identifier (VCI) in the process of connection establishment. The VCI is a part of a cell header.

Virtual Path

The virtual path (VP) represents a concept used for describing the unidirectional transport of cells belonging to virtual channels associated by a common unique identifier (VPI). It is also part of a cell header.

Figure 7.5 – The relation between Virtual Paths and Virtual Channels



The parameters of a virtual connection will be defined in the set-up phase of the connection. By use of a signalling procedure the Virtual Channel Identifier (VCI) and the Virtual Path Identifier (VPI) are requested from the local exchange. The exchanges check the availability of the requested resources and assign them if available.

The Virtual Path and the Channel Identifiers (VPI, VCI) of the header define how the cells are routed through the network. Thus, cells belonging to the same virtual connection are recognized and cells belonging to other virtual connections can be distinguished.

The values for VPI and VCI are only valid for the duration of the connection and are normally different for each link, because each switching node reprocesses them for the next link. However, they are always clearly assigned to this virtual connection. As ATM is a connection-oriented technique all cells of one connection take the same route. By this, they keep the same order; and overtaking is not possible. Preserving the cell order is known as the *cell sequence integrity principle* (ITU-T, I.150).

The VPI defines bundles of virtual connections (see Figure 7.5). Within the network, bundles of connections characterized by the same VPI can be handled together. A different application could be used in the future to distinguish different networks/operators. The subscriber could define, via the VPI-address, which network he would like to use.

In an ATM environment, a circuit or path does not have a fixed capacity. Virtual means that cells are routed from the source destination on the basis of the VPI in whatever way seems appropriate. All cells with the same VPI value belong to the same Virtual Path (VP), and a VP can contain several virtual channels. So, a VC is a logical subdivision of the VP, and all cells having the same VPI and VCI belong to the same VC.

A physical link containing VPs and VCs interconnects adjacent nodes; however the concept of VP and VC has a wider meaning than just connections from one node to another. A VP can be extended over a number of nodes. The VPI value is just a local identifier corresponding to one section of a VP. Such a section is called a *VP link*. With the help of ATM switches a series of VP links can be connected to form a complete virtual path.

Each Virtual Path can contain a number of Virtual Channels. Like the VP, the VC can be extended over a number of nodes. The section of a VC that resides within one VP is called a *VC link*. Connecting one VC link to another requires an ATM switch capable of switching on VC-level.

Virtual Path and Virtual Channel connections can be set up dynamically (on demand) through procedures initiated by the end-user, and an operator can also set-up virtual connections by use of a management system.

An example of where paths and channels could be used, would be where a company with two offices could connect their computers and telephones by renting a virtual path. In this way the company is allocated a number of virtual channels. These VCs can be used for many different services, such as e-mail, telephony, Internet access and file transfer.

As ATM is a connection-oriented technique, virtual circuits are required to be established between the end nodes before transmission can start. So, the routing of cells is performed at every node for each arriving cell. The VPI (8 or 12-bit field) together with the VCI (a 16-bit field) contain the routing information of a cell. In the routing process the VCI value of the incoming VC link is translated into the VCI value of the outgoing link.

A VP is a collection of VCs between two nodes in a B-ISDN. A predefined route is associated with each VP in the physical network. Furthermore, each VP has its own defined bandwidth, limiting the number of VCs that can be multiplexed on a VP. In general, VPIs are used to route packets between two nodes that originate, remove, or terminate the VPs, whereas VCIs are used at the end nodes to distinguish different connections.

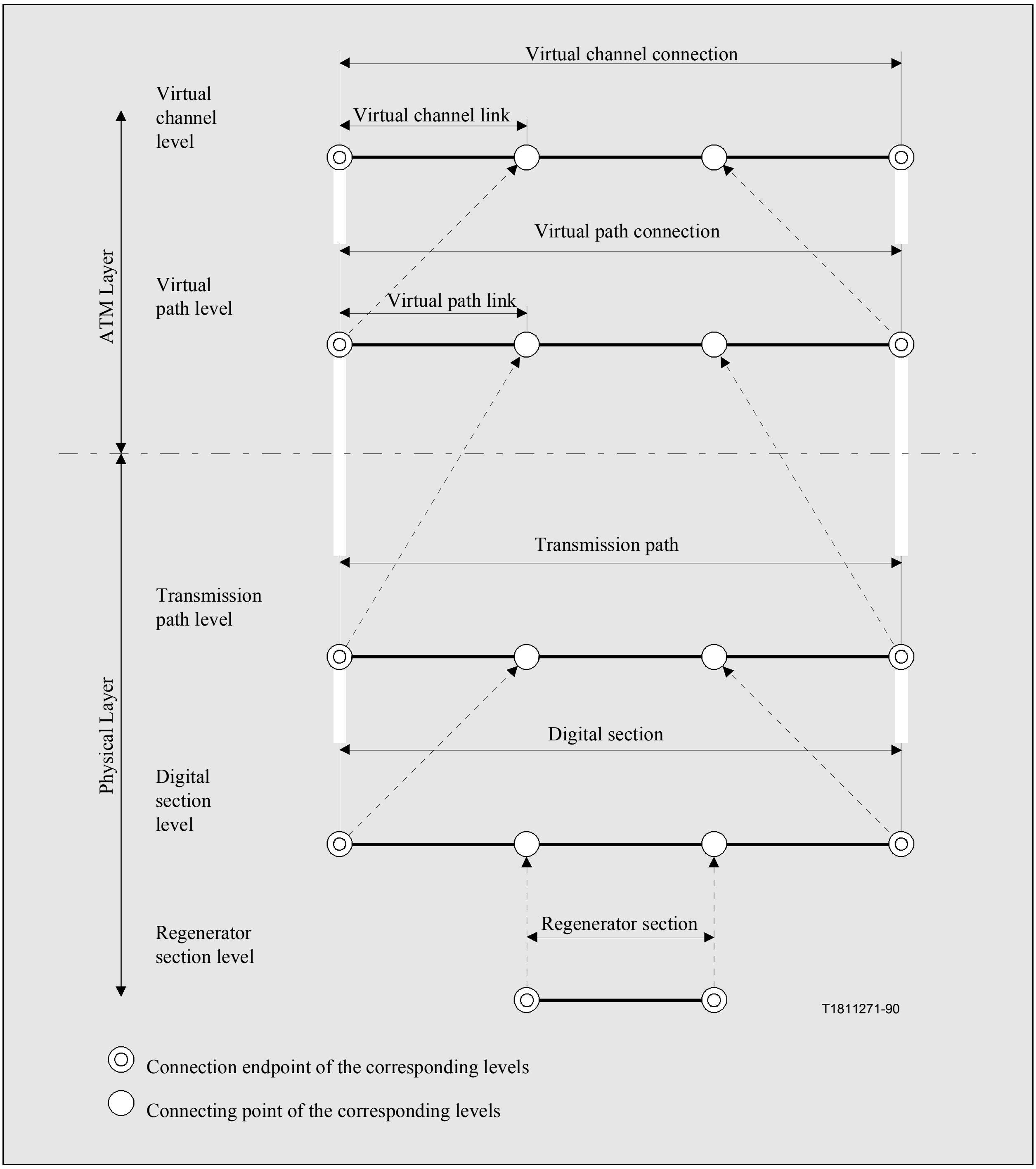
7.3 ATM transport network layering – General concept

According to ITU-T Recommendation I.311, ATM transport network is structured into two layers, the ATM layer, and the Physical layer, see Figure 7.6. The transport functions of the ATM layer are subdivided into two levels; the VC level and the VP level. The transport functions of the physical layer are subdivided into three levels, the transmission path level, the digital section level and the regenerator section level. The transport functions of the ATM Layer are independent of the physical layer implementation.

The **Connection end-point** is located at the level boundary (e.g. between the VC level and the VP level) where a client is served. The client may be located in the next higher level or in the management plane. The connection end-point provides the connection termination function.

The **Connecting point** is inside a connection where two adjacent links come together. It is located in a level where information is routed transparently. It provides the connecting function.

Figure 7.6 – Hierarchical level-to-level relationship in the ATM transport network



The **Connection** provides the capability of transferring information between endpoints. It represents the association between endpoints together with any additional information regarding the information transfer integrity.

The **Link** provides the capability of transferring information transparently. A link represents the association between contiguous connecting points or between an endpoint and its contiguous connecting point.

As shown in Figure 7.6 and Figure 7.7, a transmission path may comprise several virtual paths each of them carrying several virtual channels. The relationship between different levels of the ATM transport network is obvious.

The **Transmission path level** extends between network elements that assemble and disassemble the payload of a transmission system. At the end of a transmission path it is necessary to provide the cell delineation as well as the header error control.

The **Digital section level** extends between the network elements that assemble and disassemble a continuous bit or byte stream.

The **Regenerator section level** extends between the network elements that perform signal regeneration.

7.4 ATM switching of VPS and VCS

An ATM system permits packet switching. It uses small, equal-sized packets, called cells, and a simple protocol which allows cells to be transmitted, interpreted and delivered quickly enough to carry any kind of information including voice and video. The high bit rates require fast hardware logic for the cell processing (switching, multiplexing) instead of software processing of cells which would require more time.

The two main tasks for an ATM switching or cross-connect node are:

– the VPI/VCI evaluation and recalculation, and

– the transport of cells from input ports to the required output ports.

The switching of ATM cells is done according to a routing table. Concerning only the first switching task it can be concluded that there are two possible switching levels, the VC level and the VP level. Switching on VC level means that one incoming VC link is connected to one (or more) outgoing VC link(s). The translation is based on data in the routing table associating the incoming VPI/VCI value to the outgoing port.

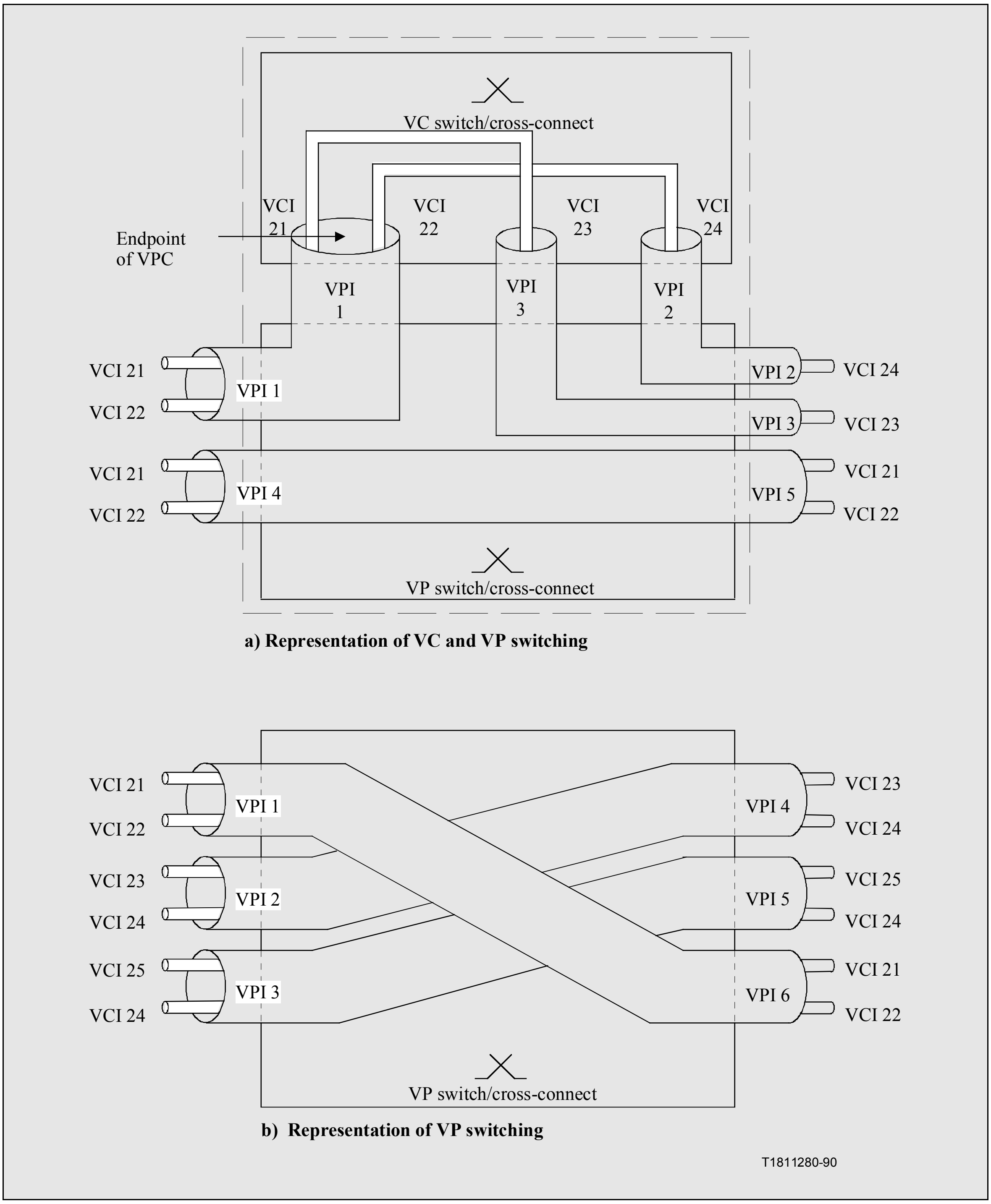
Switching on the VP level means that the routing of cells is based on the VPI value only. Thus, all incoming cells with a certain VPI value are directed to a certain outgoing VP link. No attention is paid to the VCI. The value of the VCI remains the same.

VCI identifies a particular VC link for a given Virtual Path Connection (VPC). A specific value of VCI is assigned each time a VC is switched in the network.

Routing functions of virtual channels are done at a VC switch/cross-connect. This routing involves translation of the VCI values of the incoming VC links into the VCI values of the outgoing VC links.

Virtual channel links are concatenated to form a Virtual Channel Connection (VCC). A VCC extends between two VCC endpoints or, in the case of point-to-multipoint arrangements, more than two VCC endpoints. A VCC endpoint is the point where the cell information field is exchanged between the ATM Layer and the user of the ATM layer service.

Figure 7.7 – Representing of the VP and VC switching hierarchy (ITU-T, I.311)



7.5 ATM cell format

The ATM system transports information in small packets, called cells, with

– a fixed block length of 48 bytes (octets) for the information field, and

– 5 bytes (octets) for control information (the cell header).

The advantages of the fixed size cells come from reduced queuing delay for high priority cells and efficient switching. Both of these advantages are very important for the very high data rates expected in ATM.

Figure 7.8 – Format of the ATM cell

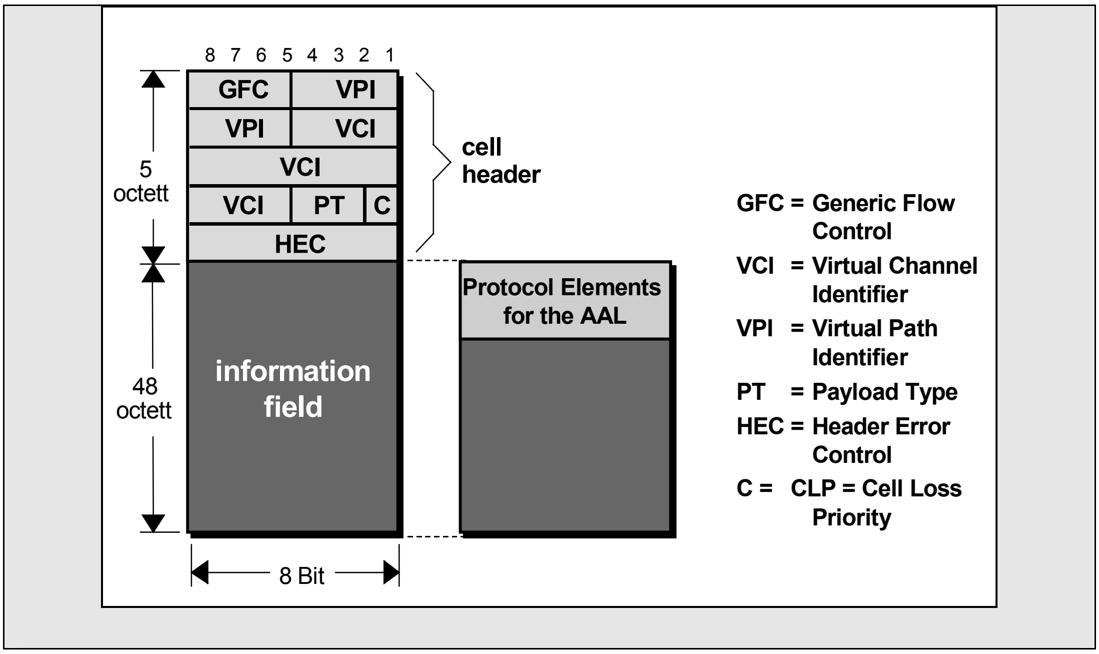


Figure 7.8 shows the structure of an ATM cell (ITU-T, I.361). This format corresponds to the cell at the user-network interface (UNI), usually the delimitation point between a public ATM network and the private installation. There is a slight difference from the cell at the network-node interface (NNI), i.e. between the nodes of the ATM network. In the NNI case, the cell header has no generic flow control (GFC) field, having an additional four VPI bits instead.

GFC – generic flow control

GFC is a four-bit field providing flow control at the UNI for the traffic originated at user equipment and directed to the network, and does not control the traffic in the other direction (i.e. for network to user traffic flow). It is not used within the network and is intended for use by access mechanisms that implement different access levels and priorities. Recommendation I.150 specifies the use of the GFC.

The GFC must be capable of ensuring that terminals access their assured capacities, especially those having elements of guaranteed capacities. The remaining spare capacity should be shared fairly among all other terminals at that UNI. There are several proposed measures for fairness. Most of them are based on equal amounts of the spare capacity, or the same percentage of the additionally requested bit rate. A comprehensive analysis of fairness at different levels is presented in a number of papers and ATM Forum contributions.

The GFC should support different delays, as well as different delay variations, and requirements that can be met on a priority basis. It should be insensitive to the traffic mix, the number of terminals, and the distance between terminals.

The main intention of ATM is to accommodate high bit rates with small delays, providing no facilities for storing cells over longer period of time. So there is no reason for GFC inside an ATM network. It controls terminals connected to a customer network.

VPI – Virtual Path Identifier Field

The VPI field at the UNI consists of eight bits (four in the first octet, and four in the second octet). These bits are used for routing. The VPI at the NNI has an additional four bits (12 in total) providing enhanced routing capabilities. The VPI is used to route cells between two nodes that originate, remove, or terminate the VPs.

VCI – Virtual Channel Identifier Field

The VCI field consists of 16 bits, and is used (together with the VPI) for routing. It distinguishes different connections and is used at the end nodes.

PTI – Payload Type Identifier

There are three bits in the ATM header to define the payload type. A value of 0 in the first bit of the PTI indicates user information, i.e. information from the next higher layer. In this case the second bit indicates whether congestion has been experienced (0 if not).

A value of 1 in the first bit is an indication that a cell carries network management or maintenance information.

CLP – Cell Loss Priority

The CLP field of the ATM cell header is a 1-bit field used explicitly to indicate the cell-loss priority. Due to the statistical multiplexing of connections, it is unavoidable that cell losses will occur in a B-ISDN connection. A cell with CLP equal to 1 may be discarded by the network during congestion, depending on the prevailing network conditions. A cell with CLP equal to 0 has higher priority and shall not be discarded if possible.

When a connection is established, the rate for higher priority cells is determined, if any (higher priority is set for services requiring a guaranteed minimum capacity).

HEC – Header Error Control

The HEC field is used mainly for two purposes: to correct (if possible) or discard cells with corrupted headers and for cell delineation. The 8-bit field provides for single-bit error correction, and a low probability that corrupted cells will be delivered. The HEC mechanism is specified by ITU-T Recommendation I.432. It is a physical layer function, capable of correcting single-bit errors and detecting multiple-bit errors in the ATM cell header.

It should be noted that the numbering convention concerning the octet positions in a cell assumes the sending of octets (bytes) in increasing order. Therefore, the cell header is sent first. Bits within an octet are sent in decreasing order, starting with the most significant bit (MSB).

7.6 The B-ISDN protocol reference model

The B-ISDN protocol reference model (B-ISDN PRM) is shown in Figure 7.9; it is composed of:

– the user plane,

– the control plane, and

– the management plane.

Above the Physical Layer, the ATM Layer provides cell transfer for all services, and the ATM Adaptation Layer (AAL) provides service-dependant functions to the layer above the AAL. The layer above the AAL in the control plane provides call control and connection control. The management plane provides network supervision functions. Functional descriptions of the Physical Layer, the ATM Layer and the AAL are given in the following section. Further study is required on the functions of the layers above the AAL.

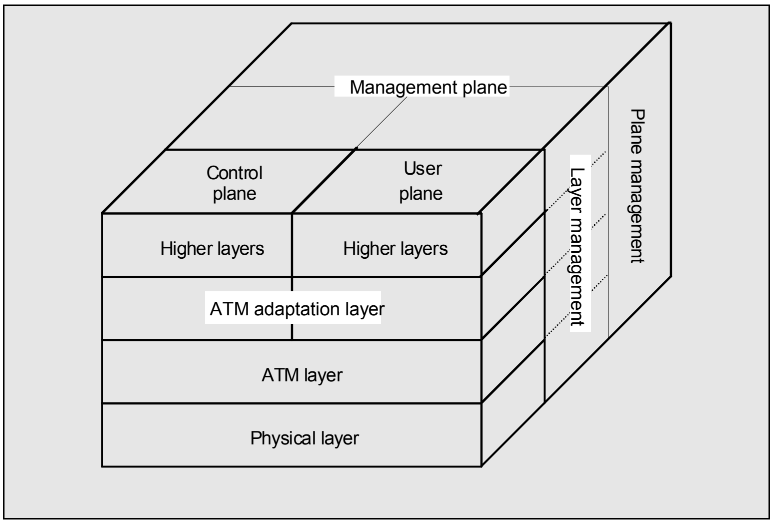
User plane

The user plane, with its layered structure, provides for user information flow transfer, along with associated controls (e.g. flow control, and recovery from errors, etc.).

Control plane

This plane has a layered structure and performs the call control and connection control functions; it deals with the signalling necessary to set up, supervise and release calls and connections. The distinction, if any, between local and global control plane functions in the broadband environment is for further study.

Figure 7.9 – B-ISDN protocol reference model (ITU-T, I.321)



Management plane

The management plane provides two types of functions, namely the Layer Management and the Plane Management functions.

a) *Plane Management functions*: The plane management performs management functions related to a system as a whole and provides coordination between all the planes. Plane management has no layered structure.

b) *Layer Management functions*: The layer management performs management functions relating to resources and parameters residing in its protocol entities. Layer Management handles the operation and maintenance (OAM) information flows specific to the layer concerned. Additional details are provided in Recommendation Q.940.

Note – A possible merger of Plane Management and Layer Management functions is for further study.

7.6.1 Functions of the individual layers of the B-ISDN PRM

The functions of each layer, the primitives exchanged between layers, and primitives exchanged between layers and the management plane are described below. The information flows described do not imply a specific physical realization. Figure 7.10 illustrates the layers of the PRM, and identifies the functions of the Physical Layer, the ATM Layer, and the AAL.

Figure 7.10 – Functions of the B-ISDN in relation to the protocol reference model

Layer management	Higher layer functions	Higher layers	
	Convergence	CS	AAL
	Segmentation and reassembly	SAR	
	Generic flow control Cell header generation/extraction Cell VPI/VCI translation Cell multiplex and demultiplex	ATM	
	Cell rate decoupling HEC header sequence generation/verification Cell delineation Transmission frame adaptation Transmission frame generation/recovery	TC	Physical layer
	Bit timing Physical medium	PM	

CS Convergence sublayer
PM Physical medium
SAR Segmentation and reassembly sublayer
TC Transmission convergence

A cell header contains only network related information. Detailed definitions describing various types of cells are given in ITU-T Recommendation I.321. Cells that carry no information concerning the ATM and all higher layers are used at the physical layer.

Idle cell – is a cell that is inserted/extracted at the physical layer in order to adapt the cell flow rate at the boundary between the ATM layer and the physical layer to the available payload capacity of the transmission system used.

Valid cell – is a cell whose header has no errors or has been modified by the HEC verification process.

Invalid cell – is a cell whose header has errors or has not been modified by the HEC verification process. It is discarded at the physical layer.

Two types of cells are passed to the ATM layer:

Assigned cell – is a cell providing a service to an application using the ATM layer service.

Unassigned cell – is a cell without assignment.

Each layer handles different aspects of the connections. Since the connections can be extended over several nodes, it is necessary to exchange information between them. On each level there are protocols to be used for that purpose.

7.6.2 Physical Layer

The physical layer is responsible for the transportation of cells. The ATM physical layer is actually divided into two parts: the physical medium (PM) sublayer and the transmission convergence (TC) sublayer.

The physical medium sublayer (PM) – is the lowest part in the PRM hierarchy and includes only the physical medium dependent functions. It is responsible for sending and receiving a continuous flow of bits with associated timing information to synchronize transmission and reception. So, the transmission capability including the generation and reception of waveforms that are suitable for the medium (insertion and extraction of timing information, line coding) is the main PM task.

The transmission convergence sublayer (TC) – is responsible for the cell delineation (the identification of the cell boundaries, I.432), header error control, cell rate decoupling, transmission frame adaptation and transmission frame generation and recovery. Transmission frame adaptation is responsible for all the actions necessary to adapt the cell flow according to the used payload structure of the transmission system in the sending direction. This frame may be cell equivalent (if no external frame is used) or dependent on the transport network (for instance SDH, xDSL).

7.6.3 The ATM Layer

The ATM layer puts cells together in the correct format. It takes the 48-byte information field formed in the AAL and adds the 5-byte ATM cell header. The VPI and VCI values for each cell are calculated in the ATM layer.

As mentioned previously the handling of Virtual Paths and Virtual Channels involves switching the cells coming in from the physical layer, based on routing tables. This routing handled by the ATM layer.

ATM layer features are independent of the physical medium. There are four main functions which this layer is responsible for:

- *Cell multiplexing* where the composite stream is a non-continuous cell flow. At the receiving side, demultiplexing into individual cell flows appropriate to VP and VC is performed.

- *VPI and VCI translation* are performed at the switching nodes and/or cross/connect nodes.

- *Cell header generation/extraction* is performed at the terminating points of the ATM layer. The information field (the 48 octets following the header) is passed to the higher layer (the adaptation ATM layer).

- *The GFC function* is defined for the user-to-network interface. It can be used to alleviate short-term overload conditions. GFC information is carried in assigned or unassigned cells.

All cells are not equally prioritised. Within the same quality of service (QoS) the priority of cells is denoted by the value of the CLP bit in the cell header. If congestion occurs the lowest priority cells are discarded first. Also the cells flowing in a VP or VC must be sent in the right order. These functions are handled in the ATM layer.

7.6.4 The ATM Adaptation Layer

The use of ATM generates the problem of supporting information flows which are not based on ATM. The simplest example is PCM (pulse code modulation), sent in a continuous stream of octets. How to assemble PCM bits into cells for transmission, and how to read them on the receiving side, is the matter for the Adaptation Layer.

ATM Service Classes

The ITU have identified five service classes in respect of bit rate, timing relation between source and receiver, and connection mode. They are listed in Table 1.

Table 7.1 – B-ISDN Service Classes, according to the ITU-T

Class	Bit Rate	Timing Relation	Connection mode	AAL protocol	Example
A stream	constant – CBR	required	connection-oriented	Type 1	voice
B stream	variable – VBR	required	connection-oriented	Type 2	video
C data	variable – VBR	not required	connection-oriented	Type 3, 4 Type 5	bursty data
D data	variable – VBR	not required	connectionless	Type 3, 4	data
X data	variable – VBR	not required	connection-oriented or connectionless	Type 3, 4	data

AAL Protocols

The ATM Adaptation Layer (AAL) maps user and signalling information into ATM cells. The AAL itself is divided into sub-layers. The AAL is involved only at the periphery of the ATM network, at the interface to the service network.

- The *AAL Convergence Sublayer* (CS) consists of a common part and a service part. It interfaces between the particular services supported and the segmentation and re-assembly sublayer below.
- The *Segmentation And Re-assembly sublayer* (SAR) carries out the conversion to and from the ATM cell format and may provide error detection and multiplexing. There are different types of SAR for different service types.

Different service applications require different characteristics from the bearer service. Some require an unspecified bit rate, for example Internet applications, while others require a constant bit rate, for example, the PSTN. Due to the compression techniques used, interactive TV can be accommodated with a variable bit rate. The AAL adapts the ATM layer to the requirements of different ATM user applications.

Several adaptation protocols are defined for the AAL. To minimize the number, the ITU-T proposed a service classification, specific to the AAL. Classifications were made with respect to the following parameters:

– the timing relation (if required),

– the bit rate (constant or variable bit rate), and

– the connection mode (connection-oriented or connectionless).

The protocols were defined for use with different traffic types (ITU-T, I.363), as outlined below:

AAL1 This is the adaptation protocol for Constant Bit Rate (CBR) services. It is intended for connection-oriented services which require a timing relationship between the source-destination pair (I.363.1)

AAL2 This is the adaptation protocol for Variable Bit Rate (VBR) connection-oriented services requiring a timing relationship between the source-destination pair (I.363.2).

AAL3/AAL4 The combined AAL3/AAL4 protocol was defined principally for linking LANs and WANs. This is the adaptation protocol for VBR, connection-oriented services which require no timing relationship between the source-destination pair (I.363.3).

AAL5 This protocol is suitable for packet data and signalling applications (both are of a VBR type). Thus, it is intended for connectionless applications without timing requirements (I.363.5).

Detailed explanations of particular AAL protocols, based on the classification of the ATM services are given in the ITU-T Recommendations (I.363.1 to I.363.5).

As the ATM-Layer provides a service-independent transport mechanism, an ATM Adaptation Layer (AAL) supports the higher layers by service dependent additional information. This information is transmitted in the information field of the cell and therefore is transparent to the ATM-Layer. In order not to define an AAL for each service, the services are divided into different classes. The functions of the AAL are different for each different class of service.

Examples of these functions are:

– segmentation of the payload in the cells,
– reconstruction of the payload from the cells,
– handling of the different cell delays,
– handling of lost and misrouted cells,
– timing reconstruction on the receiving side,
– bit error detection in the information field,
– handling of bit errors of the information field,
– multiplexing and demultiplexing of payload information,

The higher layers are unaware of the influence of the ATM principles. They deliver information to the AAL, which is responsible for the segmentation of the information stream into cells, protection means, and the hand-over of the cells to the ATM-Layer for further transfer.

The frequency of cells depends on the incoming information rate as illustrated in Figure 7.11.

ATM encapsulates all information streams from different services into cells, places them into a synchronous cell stream, and routes them across an ATM network. The cells are inserted into time slots (corresponding to the transfer time a cell) and are queued if time slots are not available. They are transferred asynchronously preserving the cell sequence integrity principle. However, cell flow limitation has to be applied to avoid overloading the network.

A successful ATM information exchange is based on defined protocols, specifying e.g. syntax, semantics, signalling and interfaces.

Some of the functions involved in an ATM transmission between ATM terminals are described below (in this case SDH is used as the transmission system):

From the send side

AAL: The payload depends on the transmitted service (constant or variable bit rate, connection oriented or connectionless data)

ATM: Addition of VC and VP addresses to header;

VC conversion in ATM switch;
VP conversion in ATM cross connect;
Multiplexing of cell streams.

PHY: Insertion of idle cells to adapt the bit rate to the SDH transmission system;

Generation of Header Error Control to permit header checking at the receiver;
Addition of transmission overhead information;
Insertion into SDH frame and STM signal transmission.

Figure 7.11 – Cell segmentation example

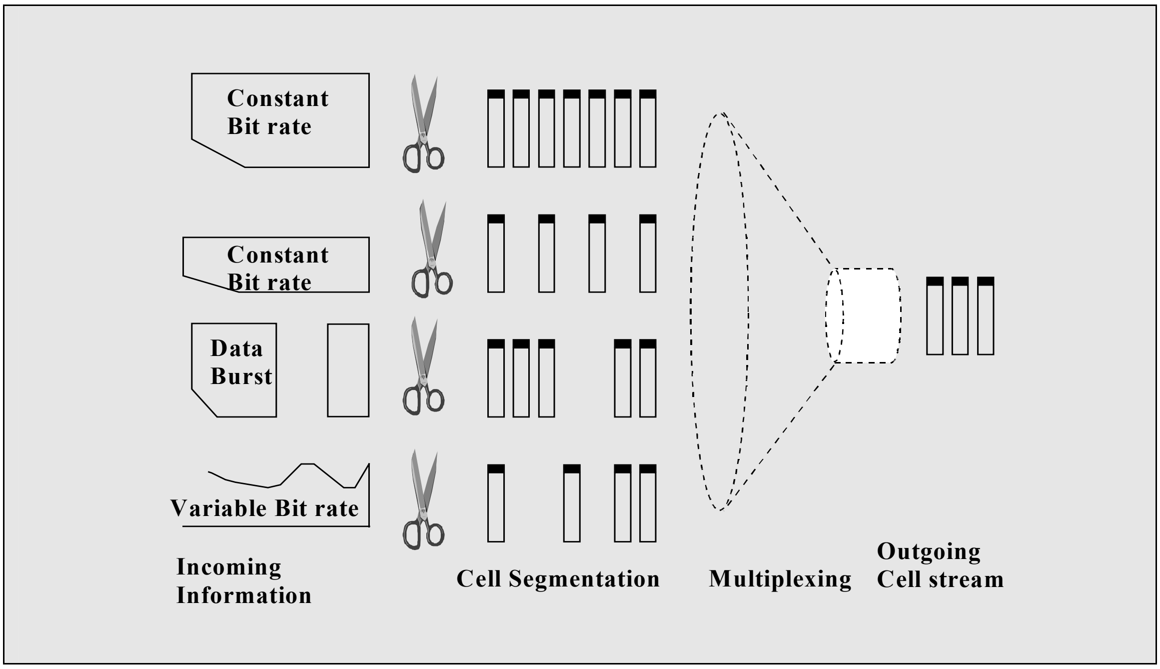
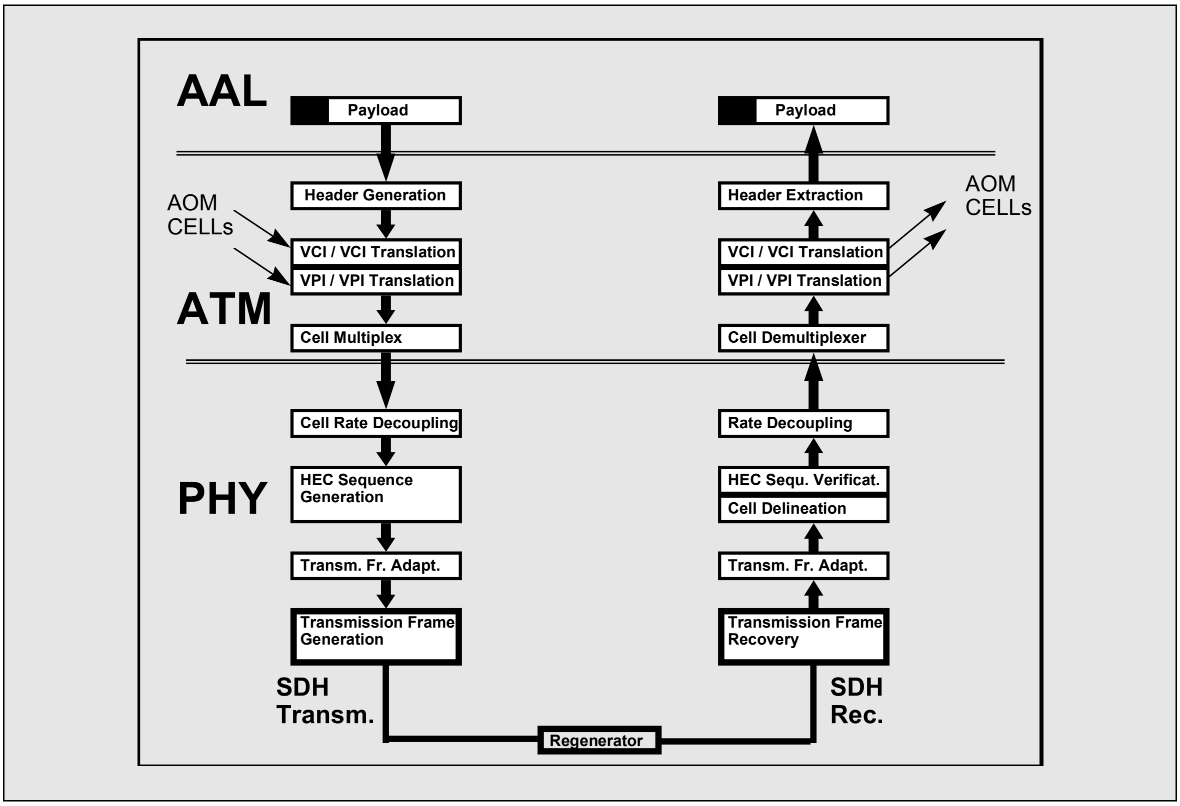


Figure 7.12 – ATM send and receive functions



To receiver side

PHY: STM signal reception and extraction from SDH frame;
Removal of transmission overhead;
Recognition of cells (cell synchronization);
Check of Header Error Control;
Elimination of cells with invalid header;
Elimination of idle cells.

ATM: demultiplexing of cell streams and transfer to their addresses;
VP conversion in ATM cross connect;
VC conversion in ATM switch;
Elimination of Header.

AAL: Delivery of the payload to the service receiver.

The header contains, in addition to the VC and VP addresses, information concerning: flow control, cell loss priority, payload type identifier and Header Error Control. In the case of overload, low priority cells are eliminated to avoid network congestion. Erroneous cells are discarded.

For end-to-end management of the channel and path, OAM cells can be inserted at the send side and retrieved at the receive side to evaluate the quality of the transmitted information.

7.7 Operation and maintenance

7.7.1 OAM principles

The following principles have to be considered in specifying the *operation and maintenance* (OAM) functions of B-ISDN.

a) *Performance monitoring*

Performance monitoring is a function which processes user information to produce maintenance information specific to the user information. This maintenance information is added to the user information at the source of a connection/link and extracted at the sink of a connection/link. Analysis of the maintenance event information at the sink of the connection allows an estimation of the transport integrity to be provided.

b) *Defect and failure detection*

Defects or failures affecting the transport of user information are detected by either continuous or periodic checking. As a result, maintenance event information or alarms will be produced.

c) *System protection*

The effect of a defect on the transport of user information is minimized by blocking or changeover to other entities. As a result the failed entity is excluded from operation.

d) *Defect information*

Defect information is given to other management entities. As a result, alarm indications are given to other management planes. Response to a status report request will also be given.

e) *Fault localization*

Determination by internal or external test systems of a failed entity if defect information is insufficient.

7.7.2 OAM levels in the B-ISDN

OAM functions in the network are performed on five OAM hierarchical levels associated with the ATM and physical layers of the protocol reference model (Recommendation I.610). The functions result in corresponding bi-directional information flows F1, F2, F3, F4 and F5 referred to as OAM flows (see Figure 7.6). Not all of these flows need to be present. The OAM functions of a missing level are performed at the next higher level. The levels are as follows:

– *Virtual channel level* (F5) – Extends between network elements performing virtual channel connection termination functions and is shown extending through one or more path connections (I.311).

– *Virtual path level* (F4) – Extends between network elements performing virtual path connection termination functions (I.311) and is shown extending through one or more transmission paths.

– *Transmission path level* (F3) – Extends between network elements assembling/disassembling the payload of a transmission system and associating it with its OAM functions. Cell delineation and Header Error Control (HEC) functions are required at the end points of each transmission path. The transmission path is connected through one or more digital sections.

– *Digital section level* (F2) – Extends between section end points and comprises a maintenance entity.

– *Regenerator section level* (F1) – A regenerator section is a portion of a digital section and as such is a maintenance sub-entity.

Relationship of OAM functions with the B-ISDN models

OAM functions are allocated to the layer management of the B-ISDN protocol reference model. This layered concept and the requirements of independence of the layers from each other lead to the following principles:

1) OAM functions related to OAM levels are independent from the OAM functions of other layers and have to be provided at each layer.

   – Each layer, where OAM functions are required, is able to carry out its own processing to obtain quality and status information. OAM functions are performed by the layer management. These results may be provided to the plane management or to the adjacent higher layer. Higher layer functions are not necessary to support the OAM of the lower layer.

7.7.3 F4 (F5) flow mechanism

The F4 (F5) flow is bi-directional. OAM cells for the F4 (F5) flow have the same VPI (VCI/VPI) value as the user cells of the VPC (VCC) and are identified by one or more pre-assigned VCI (PTI) values. The same pre-assigned VCI (PTI) value shall be used for both directions of the F4 (F5) flow. The OAM cells for both directions of the F4 (F5) flow must follow the same physical route so that any connecting points supporting that connection can correlate the fault and performance information from both directions.

There are two kinds of F4 (F5) flows, which can simultaneously exist in a VPC. These are:

– *End-to-end F4 (F5) flow* – This flow, identified by a standardized VCI (PTI) (Recommendation I.361), is used for end-to-end VPC (VCC) operations communications.

– *Segment F4 (F5) flow* – This flow, identified by a standardized VCI (PTI) (Recommendation I.361), is used for communicating operations information within the bounds of one VPC (VCC) link or multiple inter-connected VPC (VCC) links. Such a concatenation of VPC (VCC) links is called a VPC (VPC) segment.

One or more OAM segments may be defined along a VPC (VCC). Nevertheless neither overlapped nor embedded segments can be defined. For that purpose it must be ensured that all intermediate Connecting Points (CP) in between the source/sink CP of a segment shall not be a source or sink CP of another segment.

The definition of the span of a managed segment is not necessarily fixed for the duration of a connection, i.e. the managed segment may be re-configured as required.

Note – A VPC (VCC) segment is typically under the control of one Administration or organization; however, it can be extended beyond the control of one Administration/organization by mutual agreement.

End-to-end F4 (F5) flows must be terminated at the end-points of a VPC (VCC) and segment F4 (F5) flows at the connecting points terminating a VPC (VCC) segment. Intermediate points (i.e. connecting points) along the VPC (VCC) or along the VPC (VCC) segment may monitor OAM cells passing through them and insert new OAM cells, but they cannot terminate the OAM flow, except when loopbacks are performed. In this case the loopback cell may be extracted from the OAM flow by the intermediate point where the loopback has to be performed and the looped cell may be extracted by the loopback originator upon reception. The F4 flow will be initiated at or after connection set-up either by the *Telecommunication Management Network* (TMN) or by the OAM function dependent activation procedures.

A source point of a VPC (VCC) segment acting in a downstream direction should discard unexpected VPC (VCC) segment OAM cells coming from the upstream side of the connection.

It shall be possible for any intermediate connecting point to be configured as a source/sink of a VPC (VCC) segment.

7.8 Signalling in the atm network

7.8.1 Capabilities to control ATM virtual channel connections and virtual path connections for information transfer

a) Establish, maintain and release ATM VCCs and VPCs for information transfer. The establishment can be on-demand, semi-permanent and should comply with the requested connection characteristics (e.g. bandwidth, quality of service).

b) Support communication configurations on a point-to-point, multipoint and broadcast basis.

c) Negotiate the traffic characteristics of a connection at establishment of the connection.

d) Ability to renegotiate source traffic characteristics of an already established connection.

7.8.2 Capability to support multiparty and multi-connection calls:

a) Support of symmetric and asymmetric calls (e.g. low or zero bandwidths in one direction and high bandwidths in the other).

b) Simultaneous establishment and release of multiple connections associated with a call.

Note – The simultaneous establishment of multiple connections should not be significantly slower that the establishment of a single connection.

c) Two party/multiparty call establishment with or without any connections.

d) Simultaneous establishment and release of call and connection(s) associated with a call.

e) Add and release one or more connections to and from an existing call by the calling party and called party.

f) Release the call by the called party.

g) Add and release one or more a party(ies) to and from a multiparty call by the calling party.

h) Attachment/detachment of one or more parties to/from a connection.

i) Detachment from connection(s) by the called party.

j) The capability to correlate when connections compose a multi-connection call, is required.

Note – This correlation is handled by the origination and destination B-ISDN switches, which may be public or private.

k) Reconfigure a multiparty call including an existing call or splitting the original multiparty call into more calls.

7.8.3 Others

l) Capability to reconfigure an already established connection, for instance, to pass through some intermediate processing entity such as a conference bridge.

m) Support interworking between different coding schemes.

n) Support interworking with non B-ISDN services, e.g. services supported by the PSTN or 64 kbit/s based ISDN.

o) Support interaction between IN and B-ISDN.

p) Support interaction between the TMN and the B-ISDN.

q) Support failure indication and automatic protection switching for semi-permanent and permanent connections.

7.8.4 Signalling transport function

At the user access point, multiple VPs may be used to carry Signalling VCs (SVCs). These VPs may connect the user to the local exchange, other users, and/or other networks. B-ISDN signalling configurations are classified as either point-to-multipoint or point-to-point.

A point-to-multipoint signalling configuration exists when a signalling entity ("point") interacts with multiple signalling entities ("multipoint"). In a point-to-multipoint signalling configuration, meta-signalling procedures shall be used to request allocation of individual point-to-point SVCs.

A point-to-point signalling configuration exists when a signalling entity interacts with another single signalling entity.

When the signalling configuration is unknown, a point-to-multipoint signalling configuration shall be assumed. A signalling configuration can become known either by subscription or by a dynamic procedure.

7.8.5 Signalling Protocols

There are three main signalling interfaces and related protocols used in ATM networks. They are the User Network Interface (UNI), the Private Network-Network Interface (PNNI) and the B-ISDN Inter-Carrier Interface (B-ICI).

UNI Allows User Network Interface subscriber access to an ATM network, the UNI subscriber recognition through ATM addressing, and the recognition of the QoS contract and characteristics of the data to be sent across the connection.

PNNI Provides the signalling and routing protocols required for managing and controlling the ATM network configured to establish and support on-demand, switched connections and the mechanism that enable every node in the network to maintain up-to-date topological information about every other node in the network.

B-ICI The signalling and routing protocols required for managing on demand, switch connections between one ATM network and another ATM network. The B-ICI can also be used within an ATM network to improve the control of traffic routing.

7.9 Traffic management

ATM traffic management is a combination of several functions that monitor and control the information flow within the ATM network. These functions are designed to guarantee each connection its negotiated QoS. To provide this, the user must inform the network, in the connection set-up procedure, of both the expected nature of traffic as well as the type of QoS. The source node must inform the network of the traffic parameters and QoS for each direction.

A primary role of traffic management is to protect the network from congestion with the aim of complying with the network performance objectives. An additional role is realizing the efficient use of network resources.

Traffic control – represents a set of actions taken by the network in all relevant network elements to avoid congestion conditions. Congestion is considered to occur when there exists a set of one or more network elements which are not able to meet the negotiated QoS objective for the already established connections and for new connection requests. In general congestion can be caused by unpredictable statistical fluctuations of traffic flows or fault conditions within the network. Congestion control (the set of actions taken to relieve congestion by limiting the spread and duration of it) is one of the main tasks of traffic control.

To meet these objectives, the following functions form a framework for managing and controlling traffic and congestion in ATM networks and may be used in appropriate combinations. This framework is based on the fundamental concept of a *traffic contract* (the request QoS level for any given ATM connection and the maximum cell delay variation tolerance allocated to the customer equipment) that is negotiated between the user and the network, and between networks when setting up a connection.

Network Resource Management (NRM) – provisioning may be used to allocate network resources in order to separate traffic flows according to service characteristics. Although cell scheduling and resource provisioning are implementation and network specific, they can be utilized to provide appropriate isolation and access to resources.

Connection Admission Control (CAC) – is defined as the set of actions taken by the network during the call establishment phase (or during the call re-negotiation phase) in order to establish whether a virtual channel/virtual path connection request can be accepted or rejected (or whether a request for re-allocation can be accommodated). Choosing a path through the network is part of the connection admission control of the network.

ATM layer Resource Management (RM) – functions make use of resource management cells, e.g. to modify resources that are allocated to ATM connections.

Feedback controls are defined as the set of actions taken by the network and by the users to regulate the traffic submitted on ATM connections according to the state of network elements. This is related to the ABR service which uses a feedback control mechanism.

Usage/Network Parameter Control (UPC/NPC) – is defined as the set of actions taken by the network to monitor and control traffic, in terms of traffic offered and validity of the ATM connection, at the user access and the network access, respectively. Their main purpose is to protect network resources from malicious as well as unintentional misbehaviour which can affect the QoS of other already established connections by detecting violations of negotiated parameter values or procedures, and taking appropriate actions. It supervises the cell stream sent during the connection. In the case when a subscriber exceeds its booked bit rate, the *policing function* can protect the network against overload by either reducing the priority of this connection (cell tagging) or in the case of real overload, by discarding cells.

Priority controls (PC) – are functions that differentiate how cells are handled relative to each other by the network in terms of time priority or loss priority.

Traffic shaping mechanisms – may be used to achieve a desired modification to the traffic characteristics of a connection. The objectives of this function are to achieve a better network efficiency whilst meeting the QoS objectives and/or to ensure connection traffic performance at a subsequent interface.

The traffic control function is defined in the Recommendation I.371, and may be explained by the reference configuration given in Figure 7.13. It is composed of the following main components: network resource management (NRM), connection admission control (CAC), usage/network parameter control (UPC/NPC), feedback controls and priority controls. Locations of these functions are presented in the figure.

A traffic parameter is a specification of a particular traffic aspect. It may be qualitative or quantitative. Traffic parameters may for example describe:

Peak cell rate (PCR) – an upper bound on the traffic that can be submitted on an ATM connection. It is defined in terms of the variable T, the minimum spacing between cells.

Average cell rate (ACR) – the average rate of cell transfer on an ATM connection.

Sustainable cell rate (SCR) – an upper bound on the average rate of cell transfer on an ATM connection.

Cell delay variation (CDV) – the variation of actual cell arrival times on an ATM connection with respect to the theoretical cell arrival times,

Burstiness – ratio of the peak cell to the average cell rate.

Maximum burst size (MBS) – is the maximum number of cells that can be sent continuously at the peak cell rate and may be derived from service type (e.g. telephone, videophone).

Some of the above mentioned parameters are inter-dependent (e.g. the burstiness with the average and peak cell rate).

Service type can be used for implicit declaration by the user of a complete set of traffic parameters, e.g. by declaring the service requested (voice, etc.). Service type may also include implicit declaration of QoS requirements. Such a descriptor would be used for example as an address of a look-up table delivering the corresponding set of traffic characteristics. In the case where it is used by a traffic source, it would therefore not be necessary to convey any other traffic parameters belonging to the source traffic descriptor via signalling. Additionally, service type may also be used to describe traffic characteristics of a source. This applies for example when typical source behaviours (e.g. variable bit rate video using standard coding schemes) are learned from operational experience or other means and used by network operators to apply specific traffic engineering rules, which may result in QoS indications rather than commitments.

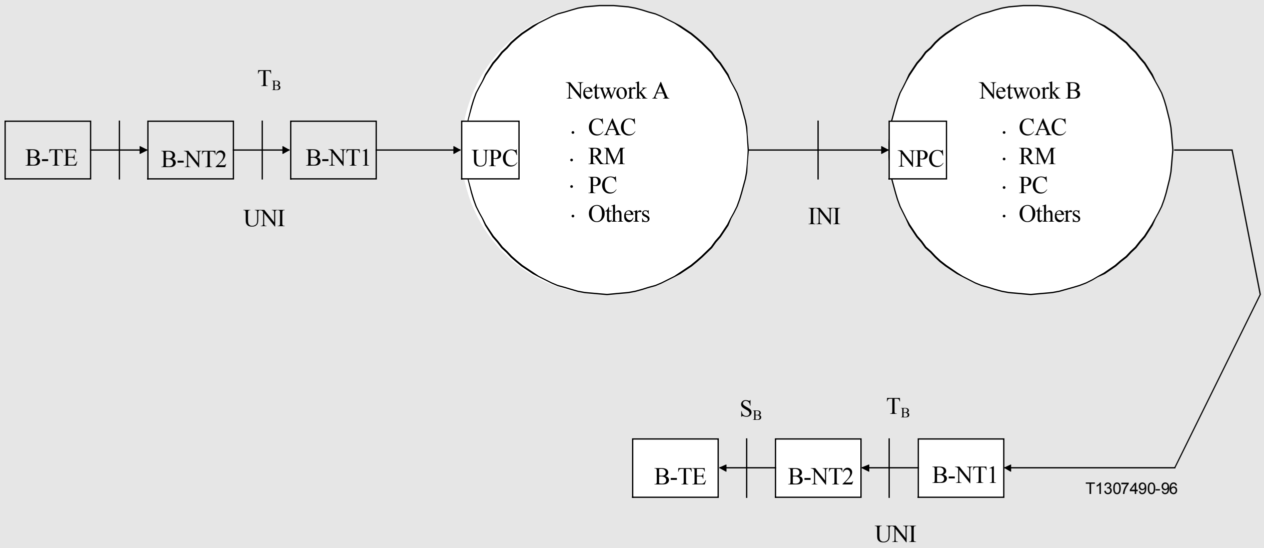
Traffic descriptor – The definition of the characteristics of the traffic that any given requested connection may offer.

ATM traffic descriptor – A generic list of traffic parameters that can be used to capture the intrinsic traffic characteristics of an ATM connection.

Source traffic descriptor – A set of traffic parameters belonging to the ATM traffic descriptor, which is used during the connection set-up to capture the intrinsic traffic characteristics of the connection requested by the source.

As a general requirement, it is desirable that a high level of consistency be achieved between the above traffic control capabilities. A specific subset of these generic functions together with relevant traffic parameters and parameter values, as well as appropriate control functions and procedures are combined to create an *ATM Transfer Capability* (ATC).

Figure 7.13 – Reference configuration for traffic control



CAC Connection Admission Control
PC Priority Control
UPC Usage Parameter Control
NPC Network Parameter Control
UNI User-Network Interface
INI Inter-Network Interface
RM Resource Management

NOTE 1 – NPC may apply as well at some intra-network interfaces.
NOTE 2 – The arrows are indicating the direction of the cell flow.
NOTE 3 – Feedback controls by means of RM cells go in the backward direction.
NOTE 4 – B-NT1 does not have any ATM layer function (see Recommendation I.413). It is included in the figure for the sake of completeness and consistency with Recommendation I.413.
NOTE 5 – In this Recommendation, UNI refers to the interface at the T_B reference point.

7.9.1 ATM Transfer Capabilities (ATC)

An **ATM Transfer Capability** is intended to support an ATM layer service model and associated QoS through a set of ATM traffic parameters and procedures.

From a user's perspective an ATC is seen as suitable for a given set of applications. The reason a user would choose a service based on a particular ATM capability other than the deterministic bit rate transfer capability, is the potential for incurring a lower cost from the network provider.

From a network operator's viewpoint, an ATC may provide gain through statistical multiplexing.

There is not a one-to-one correspondence between services or service classes (e.g. broadband bearer service categories) and ATM transfer capabilities that may be used. An upper layer data service may make use of:

Deterministic Bit Rate capability (DBR) – This capability is used by a connection that requests a particular fixed amount of bandwidth that is continuously available for the duration of the connection. This amount of bandwidth is characterized by a peak cell rate. Although this capability corresponds to CBR applications, it is not restricted only to them. This is the default ATM transfer capability.

Statistical Bit Rate capability (SBR) – In the Statistical Bit Rate (SBR) transfer capability the end-system uses standardized traffic parameters (sustainable cell rate/intrinsic burst tolerance – SCR/IBT or service type) to describe, in greater detail than just the peak cell rate, the cell flow characteristics which will be presented to the connection. It is suitable for applications where there exists some prior knowledge of some traffic characteristics of the application. The delay performance of the SBR capability is not specified, and this capability may or may not support applications with stringent delay requirements.

ATM Block Transfer capability (ABT) – An ATM Block Transfer (ABT) capability is an ATM layer mechanism for providing a service where the ATM layer transfer characteristics are negotiated on an ATM block basis. Within an ATM block accepted by the network, the network allocates sufficient resources such that the QoS received by the ATM block is equivalent to the QoS received by a DBR connection with the same peak cell rate as the negotiated peak cell rate of the ATM block, referred to as the Block Cell Rate (BCR).

Specifically, an ATM block is defined as a group of cells of an ATM connection delineated by two Resource Management (RM) cells, one before the first cell of the ATM block (leading RM cell) and the other after the last cell of the ATM block (trailing RM cell). The exact definition of the RM cells delineating an ATM block depends on the specific usage of RM cells, namely on the ABT capability. The trailing RM cell of an ATM block may be the leading RM cell of the subsequent ATM block. The BCR of an ATM block is constant over the ATM block duration.

Available bit rate transfer capability (ABR) – Many applications have the ability to reduce their information transfer rate if the network requires them to do so, or they may wish to increase their information transfer rate if there is extra bandwidth available within the network. There may be, not only static traffic parameters but also dynamic traffic parameters, as the users may be willing to accept unreserved bandwidth. To support traffic from such sources in an ATM network, an ATM transfer capability is defined which is termed Available Bit Rate (ABR).

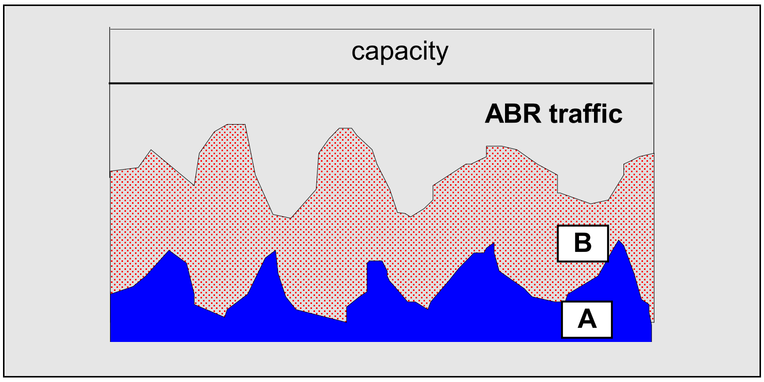
ABR is an ATM transfer capability where the limiting ATM layer transfer characteristics provided by the network may change subsequent to connection establishment. It is expected that a user that adapts its traffic to the changing ATM layer transfer characteristics will experience a low Cell Loss Ratio (CLR). Cell delay variation and cell transfer delay are not controlled. The ABR capability is not intended to support CBR applications.

The user adapts to the changing ATM layer transfer characteristics upon receiving feedback from the network. Due to cell transfer delay, this feedback reflects the status of the network at some point in time prior to the instant when the user receives it. So even when the user adapts correctly to the feedback, the network may still have to provide some buffering to enable low cell loss operation of ABR.

User actions, feedback from the network, and user responses to the feedback from the network, together constitute a control loop on the ABR connection.

A user will specify a maximum required bandwidth to the network on establishment of an ABR connection. The maximum required bandwidth is negotiated between user and network and between user and user at connection establishment. A minimum usable bandwidth (also referred to as the *minimum cell rate* or MCR) shall be specified on a per-connection basis, but it may be specified as zero. The bandwidth available from the network may become as small as the minimum usable bandwidth. The maximum required bandwidth (also referred to as the peak cell rate or PCR) and the MCR are predefined. The value of the PCR and of the MCR can be different for each direction of the connection.

Figure 7.14 – Defining of the available cell rate (ABR) service.



7.9.2 ATM Forum Service Categories

The relation of the ITU (Recommendation I.371) to the ATM Forum based on transfer capabilities is given in the af-tm-0121.000 specification. The ATM Forum defines six service categories, concerning the bit rate and specific requirements for peak cell rate (PCR), sustainable cell rate (SCR), cell delay variation tolerance (CDVT), maximum burst size (MBS), minimum cell rate (MCR), minimum frame size (MFS), cell transfer delay (CTD), cell loss ratio (CLR) and use of feedback, see Table 2.

The ATM Forum specifies classification of service categories. The first one, constant bit rate (**CBR**), is present in the ITU-T description of ATM transfer capabilities. It is used for connections that request a static amount of bandwidth that is continuously available during the connection. It is characterized by the peak cell value.

In variable bit rate transfer the ATM Forum distinguishes the two categories, the real-time (**rt-VBR**) and non-real-time (**nrt-VBR**). The differences are derived by specifying the maximum cell transfer delay requirements.

A service category with no specified bit rate (the unspecified bit rate **UBR**) is defined by the ATM Forum. It has no corresponding service capability in the ITU Recommendations. An example of such an application is traditional computer communication. The ABR and UBR services have been developed from the 'best effort' service [8].

In May 1999, the ATM Forum adopted the final version of the guaranteed frame rate (**GFR**) service intended for non-real-time applications that may require a minimum guarantee and can benefit from accessing additional bandwidth dynamically available in the network. The similar ATM service capability GFR is under study in the ITU-T.

Table 7.2 – ATM service categories according to the ATM forum

	ATM Layer Service Category					
Attribute	**CBR**	**rt-VBR**	**nrt-VBR**	**UBR**	**ABR**	**GFR**
Traffic Parameters[4]:						
PCR and CDVT[5]	Specified			Specified[2]	Specified[3]	Specified
SCR, MBS, CDVT[5]	n/a	Specified		n/a		
MCR	n/a				Specified	n/a
MCR, MBS, MFS, CDVT[5]	N/a					Specified
QoS Parameters[0]						
Peak-to-peak CDV	Specified		Unspecified			
maxCTD	Specified		Unspecified			
CLR	Specified			Unspec.	See Note 1	See Note 6
Other Attributes:						
Feedback	Unspecified				Specified	Unspec.

Notes:

1 CLR is low for sources that adjust cell flow in response to control information. Whether a quantitative value for CLR is specified is network specific.

2 Might not be subject to CAC and UPC procedures.

3 Represents the maximum rate at which ABR source may ever send. The actual rate is subject to the control information.

4 These parameters are either explicitly or implicitly specified for PVCs or SVCs.

5 CDVT refers to the Cell Delay Variation Tolerance. CDVT is not signalled. In general, CDVT need not have a unique value for a connection. Different values may apply at each interface along the path of a connection.

6 CLR is low for frames that are eligible for the service guarantee. Whether a quantitative value for CLR is specified is network specific.

7 CTD – cell transfer delay

8 MFS – maximum frame size.

7.10 ITU-T References

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7.11 Abbreviations

AAL ATM Adaptation Layer

ABR Available Bit Rate transfer capability

ABT ATM Block Transfer capability

ACR Average Cell Rate

ATC ATM Transfer Capabilities

ATD	Asynchronous Time Division
ATM	Asynchronous Transfer Mode
B-ICI	B-ISDN Inter-Carrier Interface
B-ISDN	Broadband-Integrated Services Digital Network
CAC	Connection Admission Control
CBR	Constant Bit-rate Transmission
CDV	Cell Delay Variation
CDVT	Cell Delay Variation Tolerance
CLP	Cell Loss Priority
CLR	Cell Loss Ratio
CP	Connecting Points
CS	Convergence Sub-layer
CTD	Cell Transfer Delay
DBR	Deterministic Bit Rate capability
GFC	Generic Flow Control
HEC	Header Error Control
INI	Inter-Network Interface
ISDN	Integrated Services Digital Network
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
MFS	Minimum Frame Size
N-ISDN	Narrowband-Integrated Services Digital Network
NNI	Network Network Interface
NPC	Network Parameter Control
NRM	Network Resource Management
nrt-VBR	non-real-time Variable Bit-rate Transmission
OAM	Operation and Maintenance
PC	Priority Controls
PCM	Pulse Code Modulation
PCR	Peak Cell Rate PHY Physical layer
PM	Physical Medium sub-layer
PNNI	Private Network Network Interface
PRM	Protocol Reference Model
PT	Payload Type
PTI	Payload Type Identifier

QoS	Quality of Service
RM	Resource Management
rt-VBR	real-time Variable Bit-rate Transmission
SAR	Segmentation And Reassembly sub-layer
SBR	Statistical Bit Rate capability
SCR	Sustainable Cell Rate
SDH	Synchronous Digital Hierarchy
STM	Synchronous Transfer Mode
SVC	Signalling Virtual Channel
TC	Transmission Convergence sub-layer
TMN	Telecommunications Network Management
UBR	Unspecified Bit Rate
UNI	User Network Interface
UPC	Usage Parameter Control
VBR	Variable Bit-rate Transmission
VC	Virtual Channel
VCC	Virtual Channel Connection
VCI	Virtual Channel Identifier
VP	Virtual Path
VPI	Virtual Path Identifier