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Convergence: the next big step

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CONVERGENCE: THE NEXT BIG STEP

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Purpose

The purpose of this document is to provide an M.S thesis for Masters in Telecommunications Engineering Technology at the Electrical, Computer and Telecommunications Engineering Technology department of College of Applied Science and Technology at Rochester Institute of Technology.

This thesis is a six quarter credit hour document which is offered in partial fulfillment of this degree.

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Introduction

Recently, web based multimedia services have gained popularity and have proven themselves to be viable means of communication. This has inspired the telecommunication service providers and network operators to reinvent themselves to try and provide value added IP centric services. There was need for a system which would allow new services to be introduced rapidly with reduced capital expense (CAPEX) and operational expense (OPEX) through increased efficiency in network utilization.

Various organizations and standardization agencies have been working together to establish such a system. Internet Protocol Multimedia Subsystem (IMS) is a result of these efforts.

IMS is an application level system. It is being developed by 3GPP (3^{rd} Generation Partnership Project) and 3GPP2 (3^{rd} Generation Partnership Project 2) in collaboration with IETF (Internet Engineering Task Force), ITU-T (International Telecommunication Union – Telecommunication Standardization Sector), and ETSI (European Telecommunications Standards Institute) etc. Initially, the main aim of IMS was to bring together the internet and the cellular world, but it has extended to include traditional wire line telecommunication systems as well. It utilizes existing internet protocols such as SIP (Session Initiation Protocol), AAA (Authentication, Authorization and Accounting protocol), and COPS (Common Open Policy Service) etc, and modifies them to meet the stringent requirements of reliable, real time communication systems. The advantages of IMS include easy service quality management (QoS), mobility management, service control and integration.

At present a lot of attention is being paid to providing bundled up services in the home environment. Service providers have been successful in providing traditional telephony, high speed internet and cable services in a single package. But there is very little integration among these services. IMS can provide a way to integrate them as well as extend the possibility of various other services to be added to allow increased automation in the home environment.

This thesis extends the concept of IMS to provide convergence and facilitate internetworking of the various bundled services available in the home environment; this may include but is not limited to communications (wired and wireless), entertainment, security etc. In this thesis, I present a converged home environment which has a number of elements providing a variety of communication and entertainment services. The proposed network would allow effective interworking of these elements, based on IMS architecture. My aim is to depict the possible advantages of using IMS to provide convergence, automation and integration at the residential level.

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1. Telecommunications: Brief History

1.1 Introduction

This chapter provides a brief history of the telecommunication industry. It presents various technologies and systems that were used during a particular period of time. The purpose is to highlight the advances made throughout history and depict the present state of telecommunications.

1.2 Early times: basic telephony 1880 - 1960

Telecommunication systems have gone through various changes over a long period of time. Starting from the basic voice communication in the 1880's, telephony mainly comprised of transmitting analog voice over copper lines, this lasted till about the 1950's. During this period, gradual advancements were made to improve the means of switching calls between users as the network grew. The focus was on reducing the costs, automating the system and increasing the reliability.

1.3 Digital Era: 1960- 1980

The next major change was moving from analog to digital format, which began in the 1960's, the shift was very gradual due to the high costs of transmission, but the popularity went up as the increase in the number of users caused the cost of operation to drop. The main reason for this was the drop in the prices of the digital equipment as they were being manufactured in larger numbers.

In the 1970's digital PBXs (Private Branch Exchange) were introduced; this allowed for improved and efficient call transfer with in an organization or building. During the same period out of band signaling system called Common Channel Signaling System 7 (SS7) was introduced. It was standardized by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T). SS7 signaling system increased the value of the traditional telephone network as it along with Intelligent Network (IN) allowed for enhanced services such as network based voice messaging, network based call distribution, this allows for uniform nationalized services such as 1-800 numbers. Figure 1.1 below shows a simplified diagram of a SS7 based network. It shows the centralized Enhanced Services Database residing within the SS7 network connected to the local (end office) Service Switching Point (SSP).

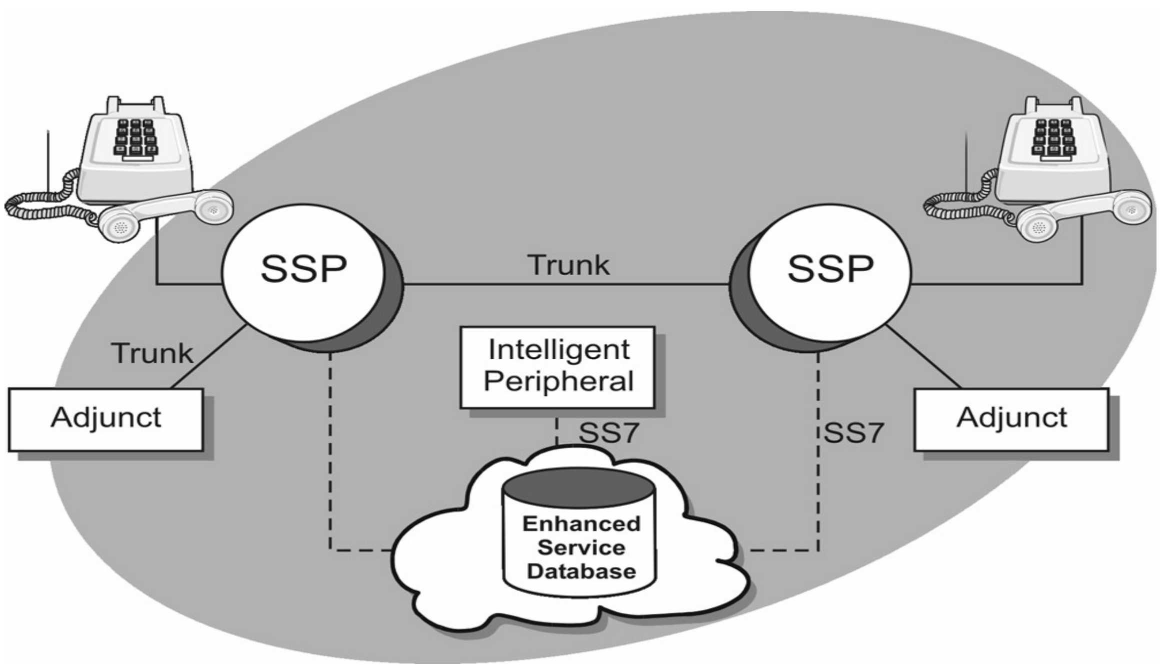

Figure 1.1 Network-based Enhanced Architecture[1]

[1] R. Dreher, L. Harte and T. Beninger , "Figure 9.1: Network-based Enhanced Services Architecture," chap. 9 in *SS7 Basics, Second Edition,* APDG Publishing 2002

We will discuss SS7 systems in more detail in the following chapters.

1.4 Data Communication: 1980- 2000

In the 1980's public mobile wireless telecommunication systems gained popularity and were able to provide viable means of communication due to the reduced costs. Even though public mobile wireless telecommunication technology was available since 1947, the market did not grow significantly until the mid 1980's. During the same time Local Area Network (LAN) technologies, especially Ethernet saw rapid growth and mass acceptance in the industry. The rate of operation of the LAN environment was 10 Mbps in the beginning, but this grew to 100 Mbps in less than a decade.

In the 1990's internet and other data communication systems and services saw major growth. The internet was created as an attempt to transfer data efficiently between research centers, but has grown to be the most popular means of communication and information exchange throughout the world. A detailed timeline of the internet is maintained by Robert H'obbes' Zakon [2]; it provides information about the various activities, key event and major technological changes that occurred over the last five decades in the internet realm. Figure 1.2 below, obtained from his website shows the increase in the number of hosts on the internet over the last 15 years.

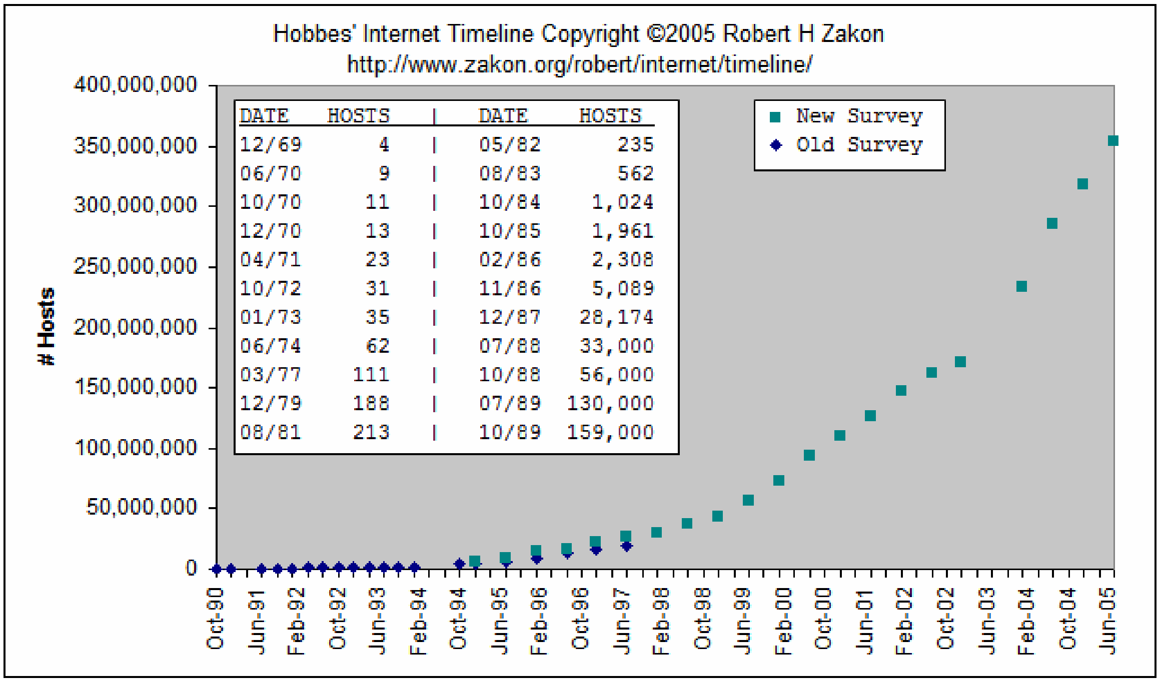

Figure1.2 Rise in numbers of hosts on the internet[2]

Figure 1.3 below shows the growth in the number of web servers in the last 15 years, it can be seen that there has been a rapid and steady growth in the numbers since 2000. This significant expansion of the internet was due to a large number of factors including the improvement in the routing and switching technologies, better long haul transport mechanisms such as ATM (Asynchronous Transfer Mode) operating at 155Mbps, SONET (Synchronous Optical Network) operating at 1Gbps or higher and other such transport technologies. These high speed transport mechanisms provided the reliable and cost effective means of communication over long distances and at high bit rates. We will discuss these systems in some depth in the following chapter.

[2] Robert H'obbes' Zakon, "Hobbes' Internet Timeline v8.1," Zakon Group LLC http://www.zakon.org/robert/internet/timeline/

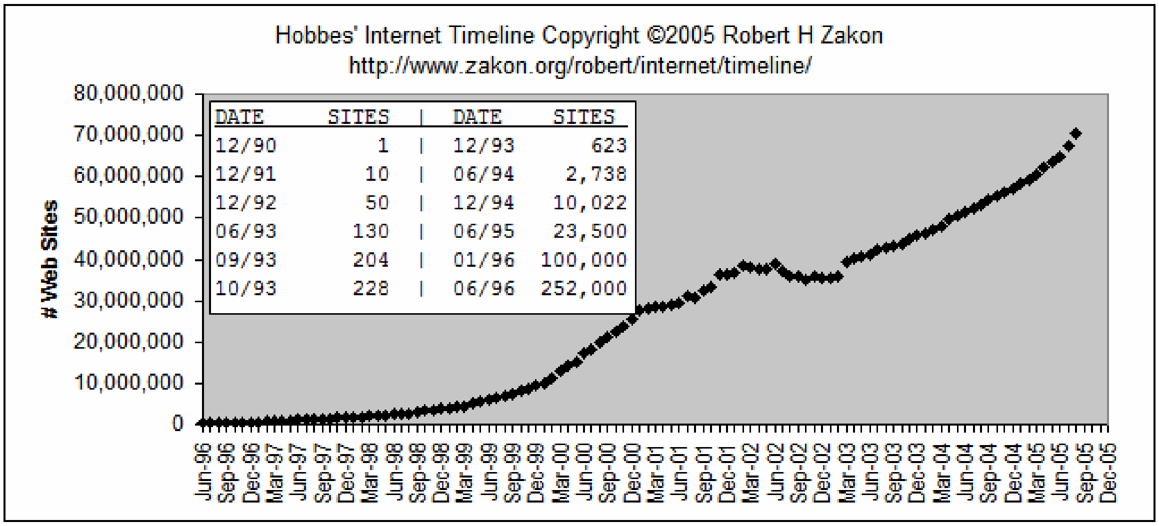

Figure1.3 Growth of the World Wide Web (WWW)[3]

1.5 Wireless Communication: 1980 - present

As mentioned earlier there were advances in the wireless communication systems, this included not just public mobile communication systems but also point to point high speed microwave communication systems; they provided transport and interconnectivity between different networks. This was an alternate to the traditional wireline based long haul transport systems.

The public mobile wireless communication has shown the most rapid advancement. It has gone through three generations in a very short period of time. The mobile or cellular communication systems are classified as 1G, 2G, 2.5G and 3G. Efforts are being taken to develop 4G wireless communication systems, but it’s still a few years away. Each of the following generations provided increased capacities in terms of distance, data transfer rate, battery life etc. The changes in the cellular systems, technologies used, and services

[3] Robert H'obbes' Zakon, “Hobbes' Internet Timeline v8.1,” Zakon Group LLC http://www.zakon.org/robert/internet/timeline/

provided will be discussed in detail in the following chapter. Figure 1.4 below gives an idea about the transition; additional information is available in [4].

Technology / Features	1G	2G	2.5G	3G	4G
Start/ Deployment	1970/ 1984	1980/ 1991	1985/ 1999	1990/ 2002	2000/ 2006
Data Bandwidth	1.9 kbps	14.4 kbps	14.4 kbps	2 Mbps	200 Mbps
Standards	AMPS	TDMA, CDMA, GSM	GPRS, EDGE, 1xRTT	WCDMA, CDMA-2000	Single unified standard
Technology	Analog cellular technology	Digital cellular technology	Digital cellular technology	Broad bandwidth CDMA, IP technology	Unified IP and seamless combination of broadband, LAN/WAN/PAN and WLAN
Service	Mobile telephony (voice)	Digital voice, short messaging	Higher capacity, packetized data	Integrated high quality audio, video and data	Dynamic information access, wearable devices
Multiplexing	FDMA	TDMA, CDMA	TDMA, CDMA	CDMA	CDMA
Switching	Circuit	Circuit	Circuit for access network & air interface; Packet for core network and data	Packet except circuit for air interface	All packet
Core Network	PSTN	PSTN	PSTN and Packet network	Packet network	Internet
Handoff	Horizontal	Horizontal	Horizontal	Horizontal	Horizontal and Vertical

Figure1.4 Transition from 1G to 4G Cellular technologies[4]

Other wireless technologies such as wireless Personal Area Network (PAN) and Wireless LAN have also gained popularity in the last 5 years; various protocols have been standardized in order to provide a uniform means of wireless communication over unregulated frequency spectrum. These protocols include WiFi (IEEE 802.11 a, b, g etc), WiMax (IEEE 802.16), Bluetooth, Zigbee (IEEE 802.15.4), RFID etc.

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4 Shakil Akhtar.: "2G-4G Networks: Evolution of Technologies, Standards, and Deployment," Encyclopedia of Multimedia Technology and Networking, Ideas Group Publisher http://faculty.uaeu.ac.ae/s.akhtar/EncyPaper04.pdf

1.6 Conclusions

The sections above provide information about the development of telecommunications; this will be helpful in understanding the following chapters which deal with the technologies being used and developed at present as well as systems required for the future. To sum it up, Figure 1.5 below shows the basic timeline of telecommunications.

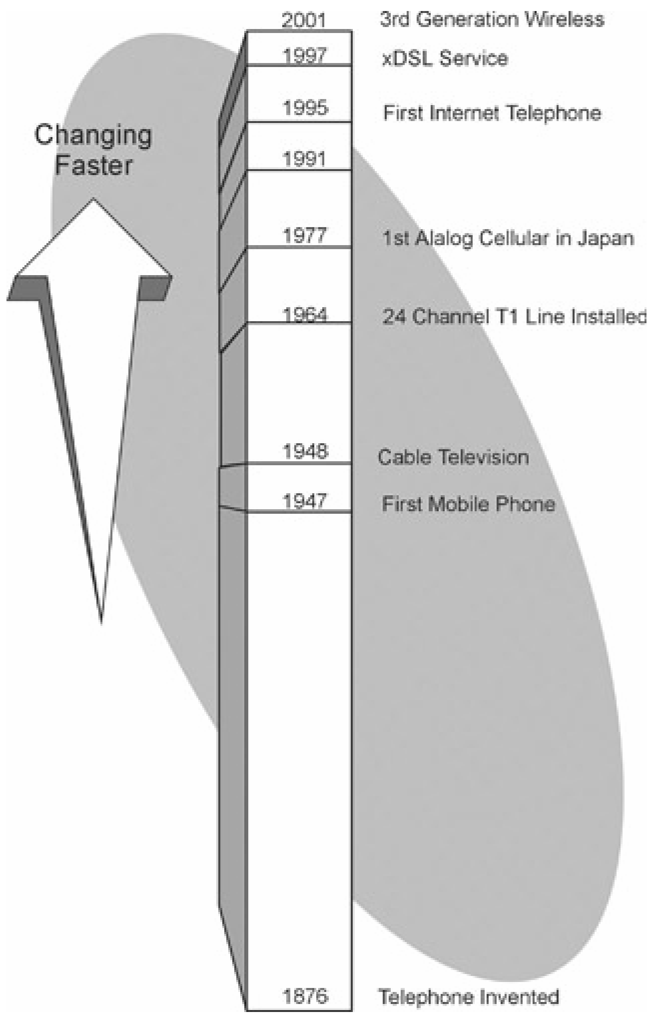

Figure1.5 Key Milestones in Telecommunication[5]

[5] Lawrence Harte, W.E. Harrelson and Avi Ofrane , "Figure 1.1: Timeline of Key Milestones in Telecommunications," chap. 1 in *Telecom Made Simple*, APDG Publishing, 2002

2. Telecommunication Systems: Present and Future Technologies

2.1 Introduction

This chapter provides information about the various telecommunication technologies being used at present and research being done to help develop the next generation of telecommunication systems. The purpose of this section is to provide the reader with sufficient background information about the technologies, wireline and wireless, their capabilities and limitations. This would make it easier to appreciate the challenges being faced in moving towards a uniform converged system. The scope of such technologies is very wide so we will restrict it to the ones that are most relevant in our discussion of convergence. Physical transmission and reception technologies will not be discussed in detail as the focus is on the application level systems of the telecommunication.

2.2 Wireline Telecommunication systems

Wireline telecommunication systems refer to all systems which use wires as a medium of transport. This includes the traditional voice and data systems such as PSTN, LAN technologies such as Ethernet, long haul technologies such as ATM, broadband internet technologies such as DSL, etc.

2.2.1 Public Switched Telephone Network (PSTN) : Voice communication

This section deals with traditional telephony, which includes voice and data communication. This is currently mostly a circuit switched environment, which allows for voice and data communication to be conducted over a twisted pair transmission cable. The network comprises a number of elements including the subscriber terminal, local loops, switches, and trunks. Figure 2.1 below shows a simplified PSTN system, the components are briefly defined below.

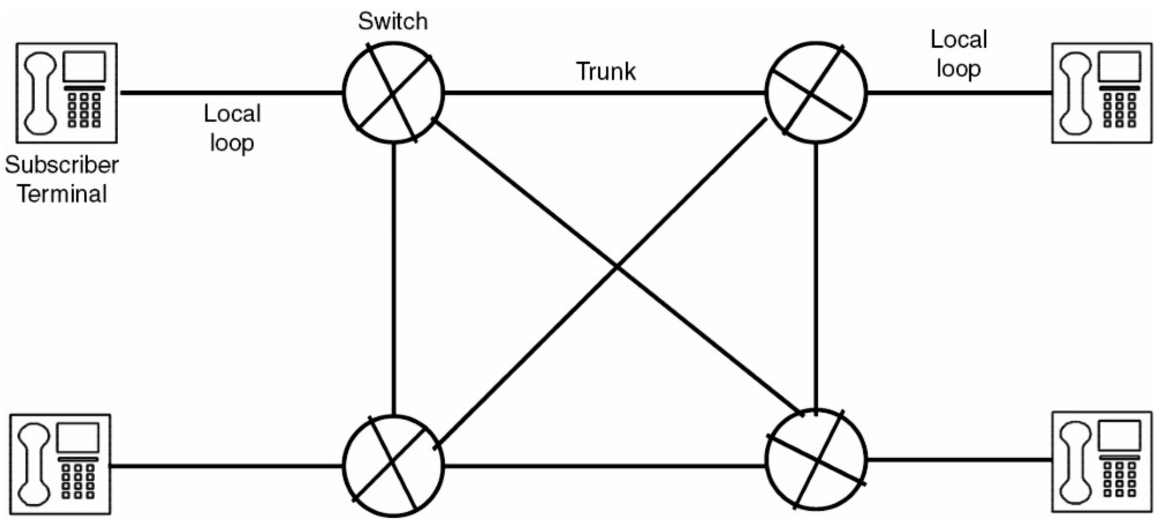

Figure 2.1 Elements of PSTN[6]

The subscriber terminal is an ordinary telephone set, capable of voice communication. The local loop is a dedicated connection between the subscriber terminal and the switch, the most popular means of providing the local loop uses twisted pair copper wire; in some remote location radio links are also used, as they prove to be more economical. Switches perform the task of setting up dedicated connections. In the past mechanical and electromechanical switches were used, but these days electronic switches perform the task; a single switch can handle thousands of calls. To provide wide coverage switches are usually organized hierarchically; a popular scheme is shown below in figure 2.2. Trunks interconnect the switches; they are high capacity lines (T1 or higher) which are

[6] Dr. K.V. Prasad , "Figure 11.1: Elements of PSTN," chap. 11 in *Principles of Digital Communication Systems and Computer Networks*, Charles River Media, 2003.

used to carry digital voice and data between exchanges[7]. The purpose of the exchange is to provide a path for a call to be completed between two parties. The primary objectives of the switch are

- Identify the subscribers
- Setup of establish communication path
- Supervise the calling process

Additional information about the telephone set, the transmission methods and systems used is provided in [6].

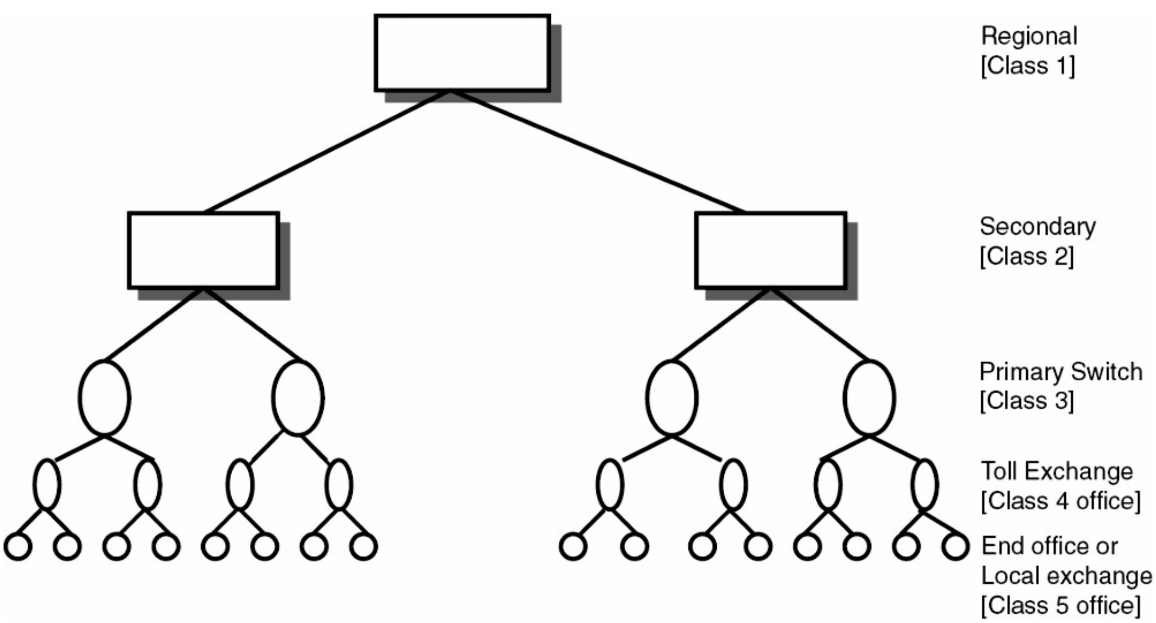

Figure 2.2 Hierarchical Switching System of PSTN[8]

Signaling is a very important part of the PSTN; in-band signaling is used to transfer information such as the telephone number to the switch. It uses the same lines. There are two other types of signaling, Channel-Associated Signaling (CAS) and Common Channel

[7] Exchanges are central locations where subscribers are interconnected. Exchanges connected directly to the local loop are called local exchanges and these are connected to the Central telephone exchange or CO.

[8] Dr. K.V. Prasad , “Figure 11.3: Hierarchical switching system of PSTN,” chap. 11 in *Principles of Digital Communication Systems and Computer Networks*, Charles River Media, 2003.

Signaling (CCS). CAS uses a different trunk line to transmit signaling information between switches, whereas CCS uses a separate communication network for information exchange. SS7 (Signaling System 7) was introduced in the previous chapter; it is used for CCS and is the most popular signaling system throughout the world.

The SS7 network consists of three main components described in[5]; Signaling Switching Points (SSP); these are telephone switches with SS7 software and terminating signaling links, the Signaling Transfer Points (STP); these are packet switches of the SS7 network. They receive signaling messages and route them to the appropriate destination and Signaling Control Points (SCP); these are databases that provide necessary information for advanced call processing capabilities such as toll-free numbers (800 numbers). Figure 2.3 below shows a simple SS7 network.

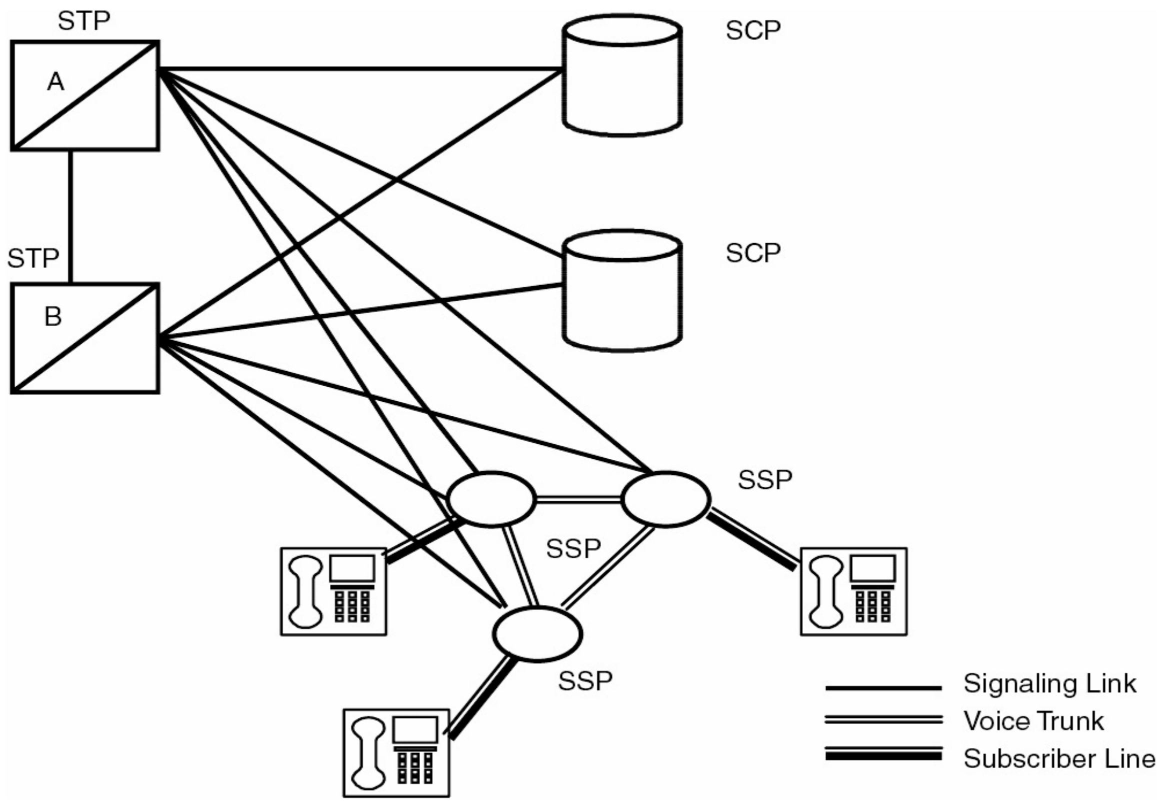

Figure 2.3 A simple SS7 network[9]

The SS7 network was designed in accordance to the OSI (Open Source Interconnect) seven layer reference model; it is defined by the ITU-T Q.7XX series recommendations. OSI was defined by the International Organization for Standardization (ISO) in 1977. The purpose of the OSI model was to promote interpretability between different systems by providing a common guideline for network data transmission between computers and components designed and manufactured by different vendors and organizations. Additional information about the OSI model can be obtained from [7].

Figure 2.4 below shows the layers of SS7 system alongside the OSI architecture.

[9] Dr. K.V. Prasad , "Figure 26.1: A smple SS7 network," chap. 26 in *Principles of Digital Communication Systems and Computer Networks*, Charles River Media, 2003.

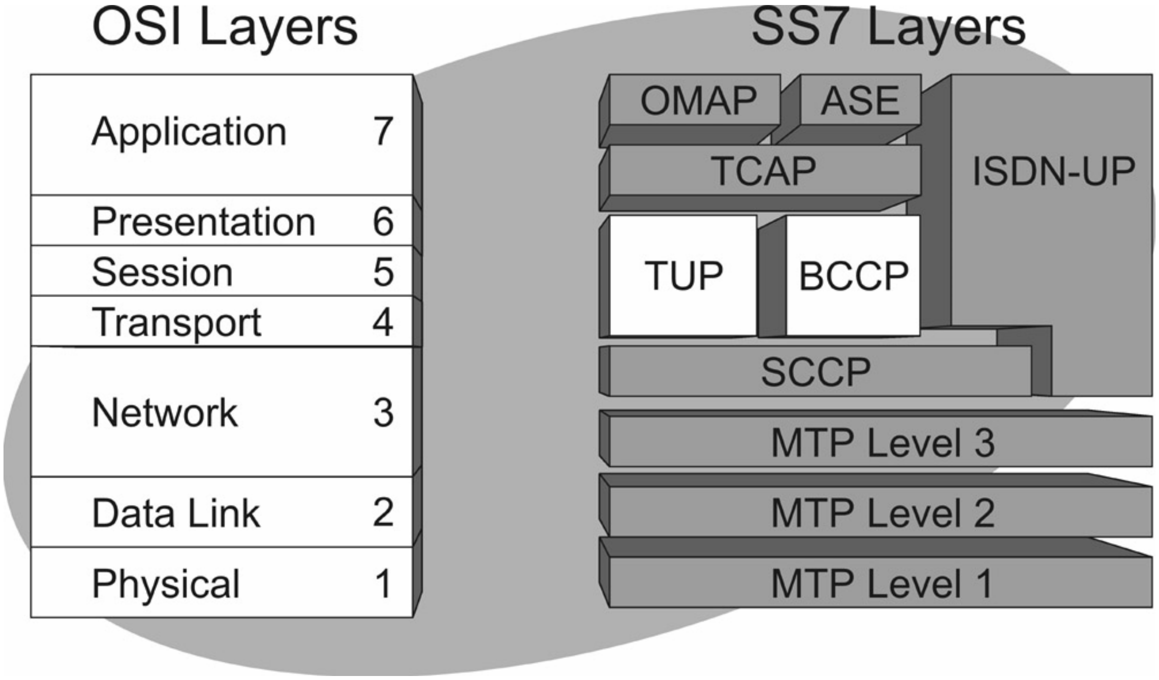

Figure 2.4 similarities between SS7 protocol and OSI model[10]

The lower part of the SS7 signaling protocol stack consists of the Message Transfer Part (MTP). It has three MTP layers (1, 2and 3) corresponding to the three lower layers of the OSI stack. The upper part of the protocol stack includes the Signaling Connection Control Part (SCCP) corresponds to the top of OSI Layer 3. The ISDN-User Part (ISDN-UP) maps onto OSI layer 3 as well, and, in addition, it maps onto Layer 4 (Transport Layer), Layer 5 (Session Layer), Layer 6 (Presentation Layer), and Layer 7 (Application Layer). The Transaction Capabilities Application Part (TCAP), the Application Service Elements (ASE), and the Operations, Maintenance and Administration Part (OMAP) of the SS7 protocol all map onto OSI Layer 7 as well. Detailed information about the different layers of SS7 can be obtained from [1].

[10] R. Dreher, L. Harte and T. Beninger , "Figure 1.4: Similarities of the SS7 Protocol and the OSI Model," chap. 1 in *SS7 Basics, Second Edition,* APDG Publishing 2002

2.2.2 Public Switched Telephone Network (PSTN) : Data communication

The PSTN generally provides data communication though one of the following means; POTS (Plain Old Telephone Service), ISDN (Integrated Digital Services Network), DSL (Digital Subscriber Line) and DLC (Digital Loop Carrier).

2.2.2.1 Plain Old Telephone Service

Data services in POTS was provided using the voice band modems. The bandwidth of the POTS cable is restricted; it allows signals within a bandwidth of approximately 300 Hz to 3000Hz. Therefore these voice modems allow only low speed communication, from 300 bps to about 56 kbps. The modems are designed and built in accordance with the ITU-T V-Series Modem Standards. The latest standard released in 2000 is the V.92; it performs asymmetric data transfer, with receive data rates up to 56kbps, and restricts the transmission data rate to 48kbps [8].

2.2.2.2 Integrated Digital Services Network

ISDN was developed as an upgrade for the traditional analog telephony system. It was a well structured all digital telephone network system. There are two interface formats available for the data transmission. The Basic Rate Interface (BRI) consisted of two channels each 64 kbps called B channel and a slower channel for data transfer and control called the D channel, its rate is 16kbps. Together they are called 2B+D channel. This was meant for residential customers for telephony and data connectivity. This system is defined by the ITU-T standard I.430 [9]. The second type of interface is called the

Primary Rate Interface (PRI); it is meant for consumers requiring high speed connectivity, mostly businesses. It comprised of a larger number of B channels depending on the local standards. In the US and Japan, the bit rate is 1.544 Mbps (T1 standard) and it includes 23B channels and 1 D channel, thus it is referred to as 23B+D. The PRI interface is defined by the ITU-T standard I.431 [10]. Figure 2.5 below shows the two interfaces.

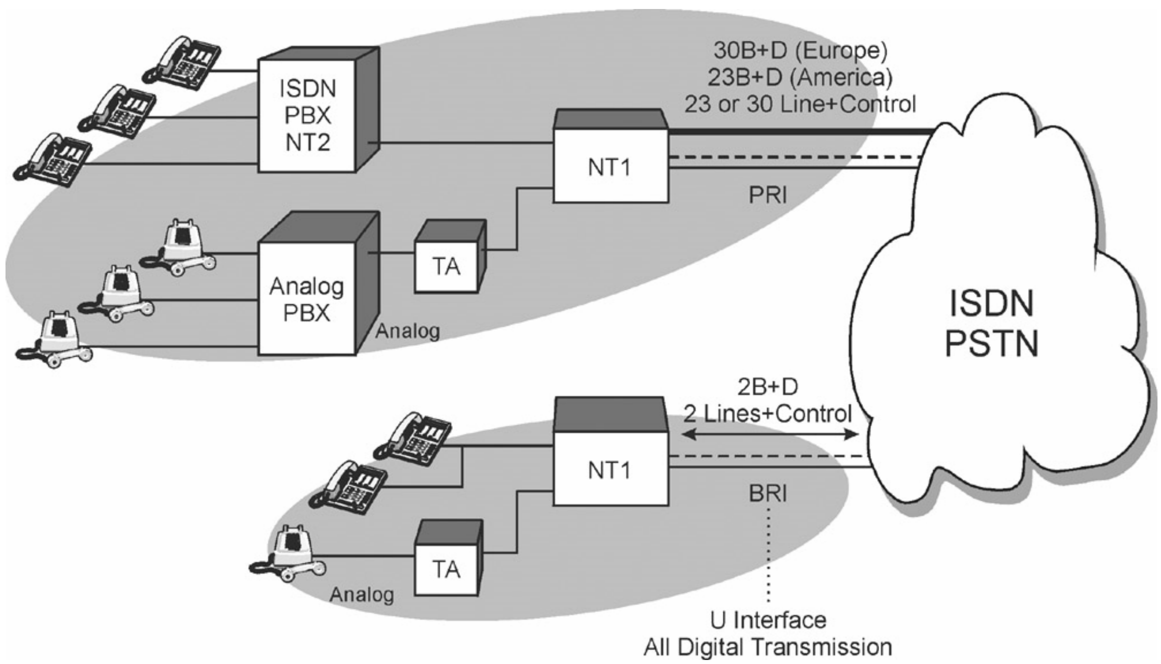

Figure 2.5 Integrated Digital Subscriber Network[11]

2.2.2.3 Digital Subscriber Line

DSL is a family of technologies that are used to provide high speed digital connection over copper wires of the traditional telephone network. It was invented at Bell Labs in 1988. It was originally designed to carry video, but the market was not ready for such an application; additional information about DSL especially ADSL is provided in [11]. In

[11] Lawrence Harte, W.E. Harrelson and Avi Ofrane , "Figure 5.8: Integrated Digital Services Network (ISDN)," chap. 5 in *Telecom Made Simple*, APDG Publishing, 2002

the mid 1990's interest developed in using DSL to deliver high speed internet traffic in the last mile.

Today ADSL (Asymmetric DSL) is capable of delivering downstream rates from 384 kbps to as high at 8 Mbps. Upstream rates vary from 64 kbps to 768 kbps. These rates depend on the length as well as the quality of the wire. DSL system uses the unutilized part of the bandwidth available in the local loop connecting the customer and the exchange. The available bandwidth is divided into channels 4312.5 Hz wide and each channel is evaluated for usability like in the analog modem. The larger the number of channels, higher the available bandwidth; this is the reason for the bit rate being dependent on the length of the wire. The channels are classified into upstream and downstream channels and virtual circuits are formed. The DSL transceivers monitor these channels for quality and add or drop channels as per need. Figure 2.6 below shows the various components of a DSL network. The customer premises equipment converts the analog and digital signals into high speed DSL signal and transmits it via a DSL modem. The local loop transports this signal to the exchange where it split into the digital data and analog voice signal. The voice information is forwarded to the POTS network and the digital data goes to the DSLAM (Digital Subscriber Line Access Module), where it is combines various digital streams and forward it to the appropriate service provider. Functions of the other components are described in [3].

There are various types of DSL systems present and being used, ISDN is also considered a DSL technology. The most popular system is ADSL (Asymmetric DSL). The difference in the technology lies in the modulation (line coding) used, data rates in the upstream and downstream path, etc.

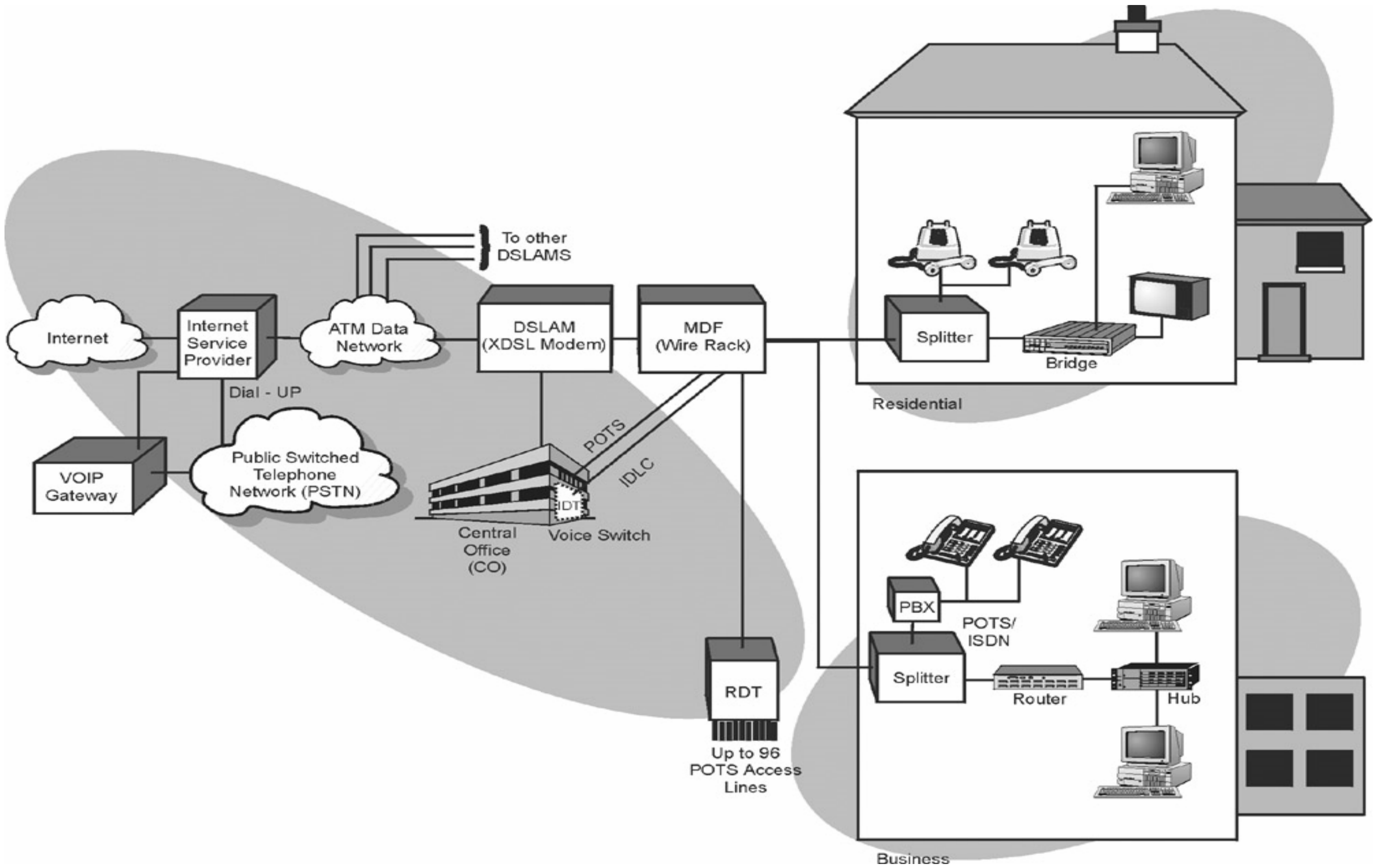

Figure 2.6 DSL Network Diagram[12]

ADSL is defined by ANSI (American National Standards Institute) standard T1.413 - 1999 [12] and by ITU-T (International Telecommunication Union) Telecommunication Standardization Sector Recommendations G.992.1 [13] and G.992.2 [14].

ADSL2 is an enhancement over ADSL in terms of the data rates and reach performance. It also provides for rate adaptation and power management. It is defined by the ITU-T recommendations G.992.3 [15] and G.992.4 [16]. ADSL2plus is a further enhancement as it doubles the bandwidth used for downstream data transmission. It achieves data rates

[12] Lawrence Harte, W.E. Harrelson and Avi Ofrane , "Figure 5.9: DSL Network Diagram," chap. 5 in *Telecom Made Simple*, APDG Publishing, 2002

of up to 20Mbps on lines as long as 5000 feet. It is defined by the ITU-T recommendation G.992.5 [17].

Other DSL technologies include HDSL (High bit rate Digital Subscriber Line), its standardized version is called SDSL (Symmetric Digital Subscriber Line), defined by ITU-T recommendation G.991.1 [18]. It used Pulse Amplitude modulation for line coding and had symmetric bit rates in the range of 784 – 2320 Kbps. HSDSL (Single-pair High-Speed Digital Subscriber Line) was an enhancement over HDSL as it provided symmetrical bit rates from 192 kbps to 4.6 Mbps. It was standardized by the ITU-T recommendation G.991.2 [19].

There are other DSL systems such as VDSL (Very high data rate DSL) and VDSL2 (Enhanced VDSL), RADSL (Rate Adaptive DSL) and CDSL (Consumer DSL). Figure 2.7 below gives a good comparison between the technologies.

SERVICE	EXPLANATION	DOWNLOAD	UPLOAD	MODE OF OPERATION
ADSL	Asymmetric DSL	1.5 to 8.192 Mbps	16 to 640 Kbps	Different up and down speeds, one pair wire.
RADSL	Rate Adaptive DSL	64 Kbps to 8.192 Mbps	16 to 768 Kbps speeds	Different up and down. Many common operations use 768 Kbps. One pair wire.
CDSL	Consumer DSL	1 Mbps	16 to 160 Kbps	Now ratified as DSL-lite (G.lite). No splitters. One pair wire.
HDSL	High-data rate DSL	1.544 Mbps in North America	1.544 Mbps	Symmetrical services. Two pairs of wire.
		2.048 Mbps	2.048 Mbps in rest of world	
IDSL	ISDN DSL	144 Kbps (64+64+16) as BRI	144 Kbps (64+64+16) as BRI	Symmetrical operation. One pair of wire. ISDN BRI
SDSL	Single DSL	1.544 Mbps, 2.048 Mbps	1.544 Mbps, 2.048 Mbps	Uses only 1 pair but typically provisioned at 768 Kbps. One pair wire.
VDSL	Very High data rate DSL	13 to 52 ± Mbps	1.5 to 6.0 Mbps	Fiber needed and ATM probably used
SHDSL (G.SHDSL)	Single High-speed DSL or 384 Kbps	192 Kbps to 2.360 Mbps or 384 Kbps to 4.720 Mbps	192 Kbps to 2.360 Mbps Using 2 pair. to 4.720 Mbps	Using 1 pair.

Figure 2.7 Different DSL services, speeds and operation[13]

[13] Regis J. "Bud" Bates, “Table 16-2: Summary of DSL speeds and operations using current methods,” chap. 16 in *Broadband Telecommunications Handbook, Second Edition*, McGraw-Hill © 2002

2.2.3 Cable Network

Cable TV operations began in the 1940's; the main purpose was to distribute RF (Radio Frequency) signal over cables rather than over the air. It gained popularity in the 1960s. It was called the Community Antenna Television (CATV). The system used 6 MHz of the electromagnetic spectrum for each television channel. In the 1970's cable operators expanded programming by televising satellite broadcasts and introducing pay per view services.

In the early 1980's the cable network was upgraded to HFC (Hybrid Optical Fiber Coaxial Cable), which allowed for increased capacity and lager coverage. The HFC system can extend up to 50 miles from the head end and carry frequencies as high as 860 MHz. The Telecommunications Act of 1996 allowed for cable companies to provide data and telephony over the cable network. Figure 2.8 below shows the upgraded hybrid cable network. At present the typical data transmission rates are about 27 Mbps downstream and 10 Mbps upstream. The operation of a hybrid system is described in [20]

In the early 1990's cable operators formed a number of research groups in order to explore the possibility of transmitting data, and MCNS (Multimedia Cable Network System) was the first one to come up with a standard. They released DOCSIS 1.0 (Data-Over-Cable Service Interface Specification) in March 1997. At present CableLabs, a non profit research and development consortium, working in collaboration with MCNS, is responsible for developing new specifications and product definitions.

The DOCSIS [22] specification describes DOCSIS as a tree based network with the Cable Modem Termination System (CMTS) as the root of the tree and the Cable Modem (CM) as the leaves. The CMTS is located at the service provider location and the CMs are located at the residential users' home. The transmission of data from the CMTS to the CMs is a point to multipoint broadcast and the transmission from the CM to the CMTS (upstream) is controlled by the CMTS and is a multipoint to point TDMA (Time Division Multiple access) process. Typically there are 1500 to 2000 CMs connected to a single CMTS.

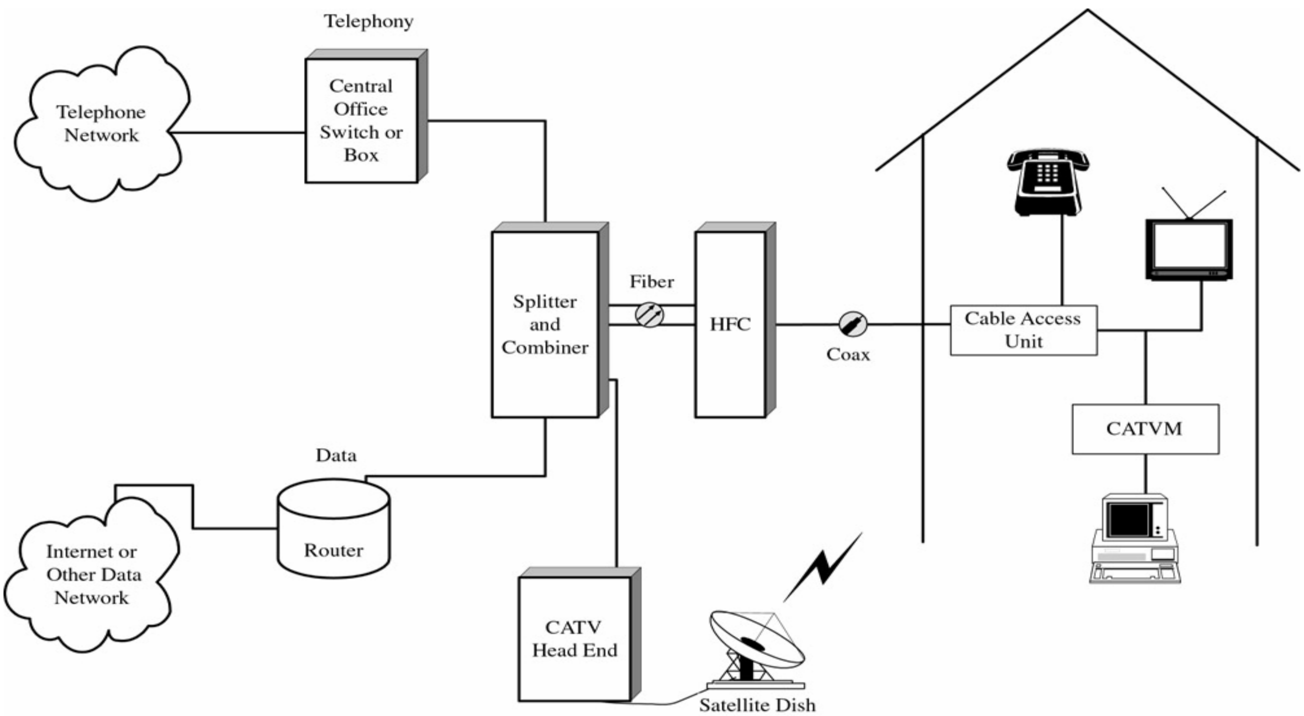

Figure 2.8 Hybrid cable data network[14]

To date two other DOCSIS specifications have been released by CableLabs. DOCSIS 1.1 was released in 1999; it provided enhanced security, added Quality of Service (QoS) to support real- time applications. DOCSIS 2.0 was released in 2001; it provided significant upstream capacity upgrade to allow high speed applications. Information about all the

[14] Regis J. "Bud" Bates, "Figure 15-4: The new hybrid data network," chap. 15 in *Broadband Telecommunications Handbook, Second Edition*, McGraw-Hill © 2002

specifications and other projects being undertaken by CableLabs is available on their website [23].

DOCSIS conforms to the OSI model of network architecture. Figure 2.9 below shows the same.

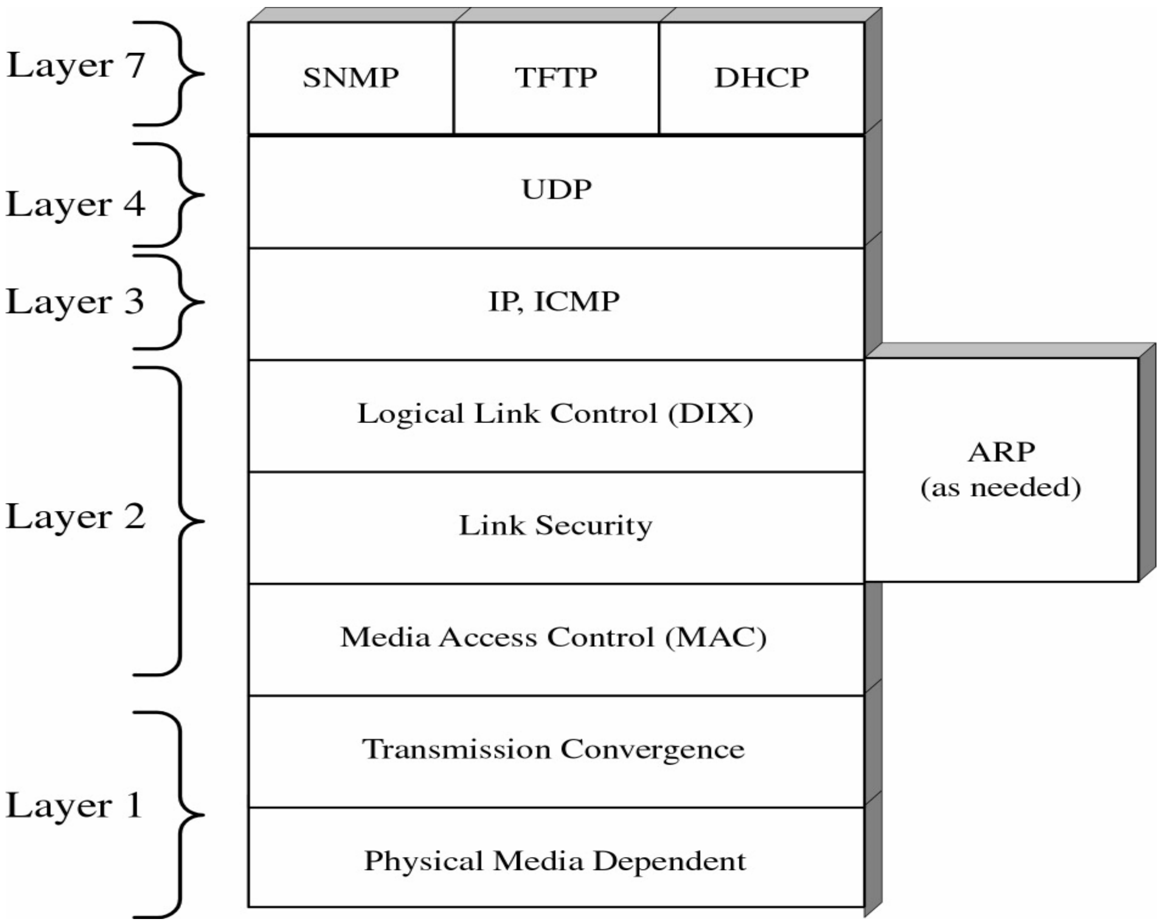

Figure 2.9 DOCSIS layers[15]

The cable network configuration supports information transport at Layer 1 of the OSI Reference Model or the Physical Layer. For Layer 2 or the Data-Link Layer, cable networks employ the Medium Access Control (MAC) protocol to facilitate collision detection and retransmission, error detection and error recovery, and timing and

15 Regis J. "Bud" Bates, “Figure 15-6: The DOCSIS model,” chap. 15 in *Broadband Telecommunications Handbook, Second Edition*, McGraw-Hill © 2002

synchronization functions. At the Network Layer or Layer 3, wireline cable modem configurations enable transmission of IP (Internet Protocol) packets via the HFC infrastructure. At the Transport Layer or Layer 4, cable networks operate in conjunction with the User Datagram Protocol (UDP) and the Transmission Control Protocol (TCP). At the upper layer of the cable modem protocol stack, cable networks employ SNMPv3 (Simple Network Management Protocol version 3) for managing and administering network operations.

2.2.4 Local Area Network

A Local Area Network (LAN) can be defined as a collection of computers and peripherals networked together by some kind of a standard system. Different type of media as well as configurations can be used to network these peripherals. These networks usually operate over a small area, a building, or a campus. The networks are designed using various network topologies, such as bus, star, ring and mesh topologies. Each of them has their advantages and disadvantages; they are suited for different networks with different requirements.

The requirements of a network include parameters such as number of nodes, data rates required, type of applications, media used etc. There are three LAN technologies which have become most popular; they are Ethernet, token ring, and Fiber Distributed Data Interface (FDDI).

2.2.4.1 Token Ring

The Token ring network is a token-passing ring shaped LAN. It was developed by IBM and operates at 4 Mbps. It supports 72 devices on standards telephone cable. Using shielded twisted-pair wiring, the network supports up to 260 devices. Although it is based on a ring (closed loop) topology, the Token Ring network uses star-shaped clusters of up to eight workstations connected to a wiring concentrator Multi Station Access Unit (MSAU), which, in turn, is connected to the main ring. The Token Ring network is designed to accommodate microcomputers, minicomputers, and mainframes; it follows the IEEE 802.5 standard [24]. Figure 2.10 shows a typical token ring network which uses the IBM architecture described in [25].

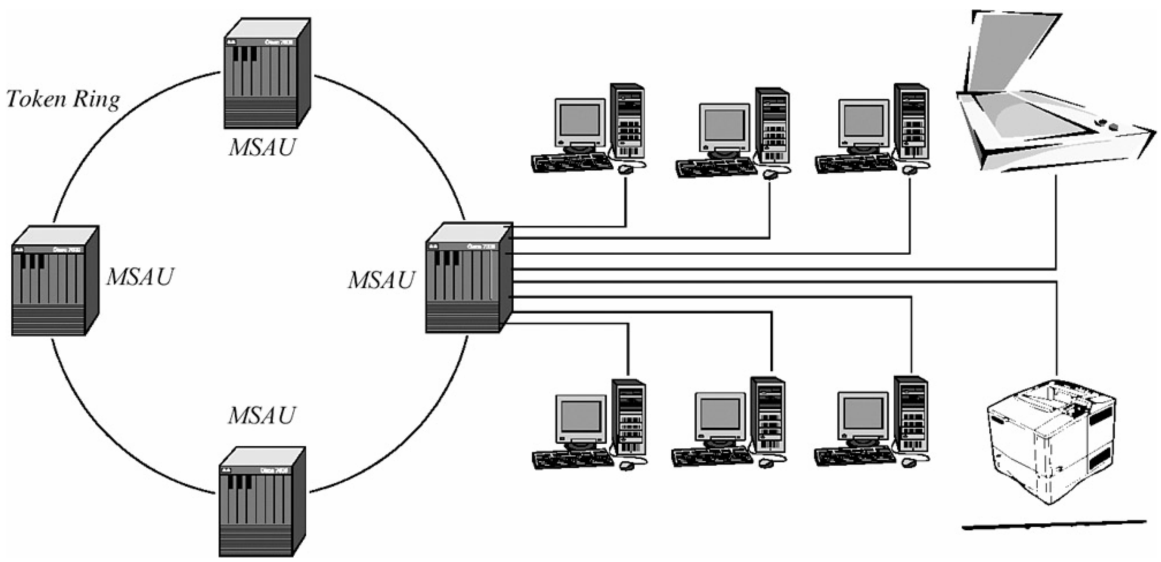

Figure 2.10 Token ring LAN environment[16]

2.2.4.2 Fiber Distributed Data Interface

Fiber Distributed Data Interface (FDDI) is a 100 Mbps LAN technology; it is defined by ANSI standard X3T9.5. It is similar to token ring system as it uses token passing but it

[16]Stephen T. Karris, “Figure 4.14: IBM Token Ring with MSAUs,” chap. 4 in Networks: Design and Management, Orchard Publications, 2004.

uses fiber optic cable; also, it allows for several devices to transmit at once. Instead of using hubs like in token ring, FDDI uses concentrators[17] to connect devices.

FDDI uses a topology referred to as dual attached, counter rotating token ring. In this topology the device is connected to both FDDI token passing rings; it allows for uninterrupted service in case of failure of either of the rings. Figure 2.11 below, shows the dual ring connection architecture of the FDDI network.

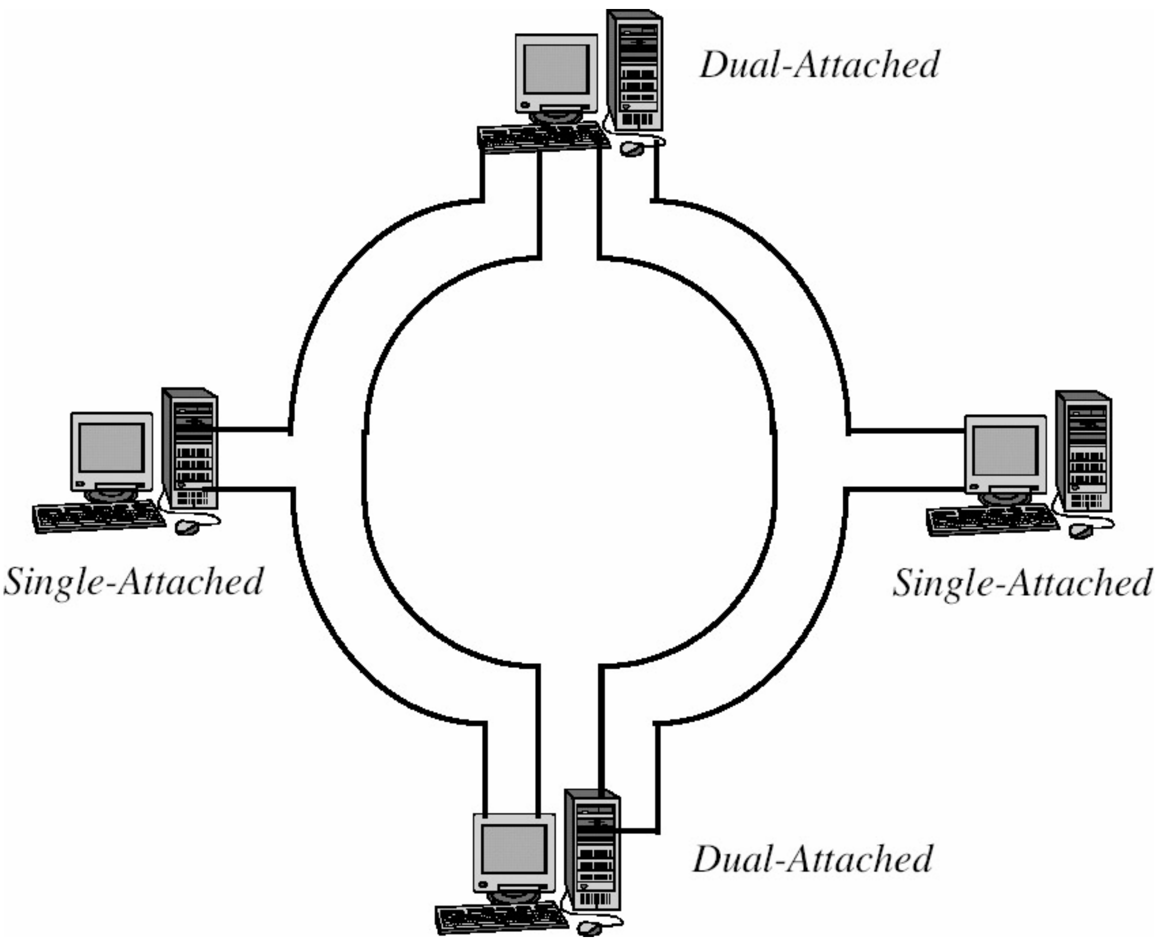

Figure 2.11FDDI network dual link architecture[18]

[17] A concentrator is a type of multiplexer that combines multiple channels onto a single transmission medium in such a way that all the individual channels can be simultaneously active. For example, ISPs use concentrators to combine their dial-up modem connections onto faster T1 lines that connect to the Internet.

[18] Stephen T. Karris, "Figure 4.21: Typical connection of a Dual-ring FDDI network," chap. 4 in Networks: Design and Management, Orchard Publications, 2004.

2.2.4.3 Ethernet

Ethernet is the most widely used LAN technology. Over 100 million Ethernets have been installed worldwide; the main reason for the popularity is scalability. The Ethernet technology suite is defined by the IEEE 802.3 family of standards [26] and it annexes. Ethernet platform inter works with technologies that include Frame Relay (FR), FDDI (Fiber Data Distributed Interface), Fiber Channel (FC), DSL (Digital Subscriber Line), cable modem, WDM (Wavelength Division Multiplexing), and DWDM (Dense WDM).

Figure 2.12 below shows an Ethernet network. At present Ethernet systems are capable of running at speeds of 10 Mbps, 100 Mbps, 1000 Mbps or 1 Gbps, and 10000 Mbps or 10 Gbps via diverse media. A typical Ethernet system consists of physical components that include network stations and nodes; interface devices such as hubs, switches, routers, bridges; repeaters for regenerating Ethernet signal and Network Interface cards (NIC) to connect the nodes to the network.

Ethernet operates on the lower layers of the OSI stack. It operates at layer 1 or the physical layer and layer 2 or the data link layer. The Data-Link Layer or Layer 2 is responsible for transmission and reception of Ethernet frames, specifies procedures for accessing the Physical Layer, and employs multiplexing techniques such as CSMA/CD for effective transmission. The Data-Link Layer includes the Media Access Control (MAC) Sub layer and the Logical Link Control (LLC) Sub layer. At the Data-Link Layer, all the four types of Ethernet (Ethernet, Fast Ethernet, Gigabit Ethernet, and 10 Gigabit Ethernet) employ the MAC Sub layer for translating data into Ethernet frame formats; this is explained in [21].

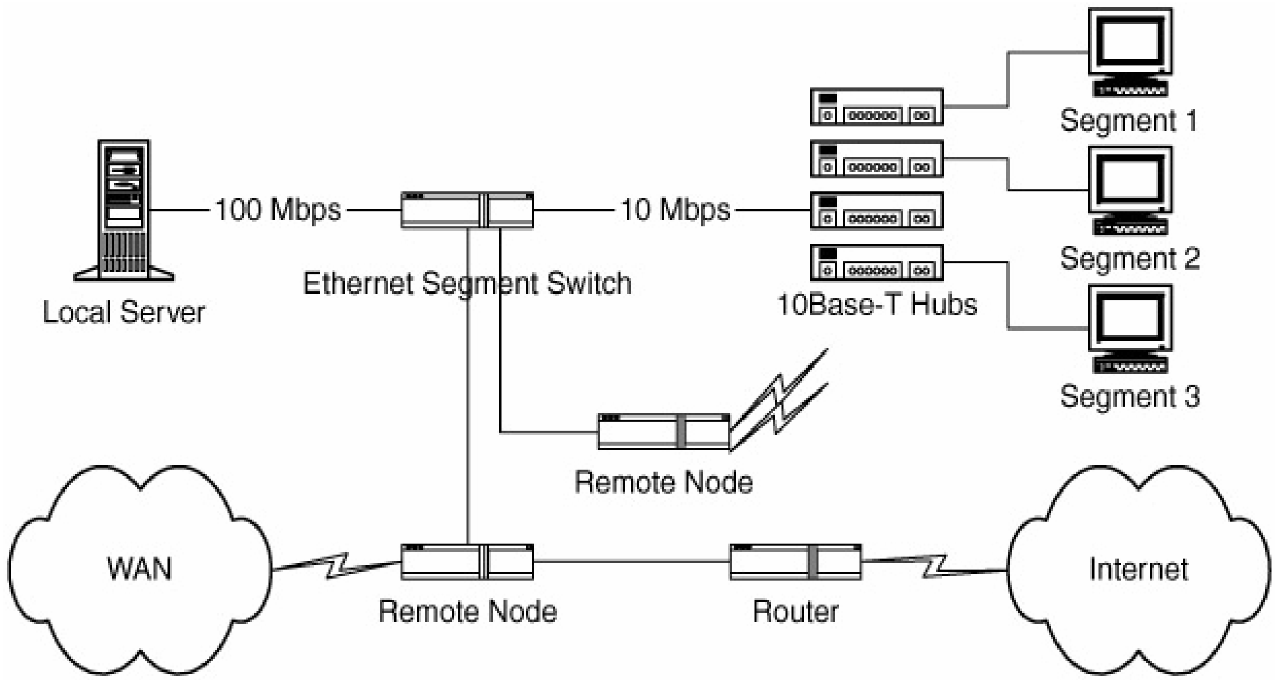

Figure 2.12 A Ethernet/Fast Ethernet LAN configuration with multiple access points[19]

Ethernet can be classified according to the speed of operation, the type of media used or the topology implemented. 10 Mbps Ethernet systems can be implemented using various media. These media include thick coaxial cabling for 10BASE-5; 5 refers to the maximum length of segment in this case 500 meters. Thin coaxial cable is used for 10BASE-2; the maximum length of segment is 200 meters. Unshielded Twisted Pair (UTP) copper wiring is used for 10BASE-T; the maximum length of the segment is 100 meters. Optical fiber is used for 10BASE-F and 10BASE-F sub-categories; specifically, 10BASE-FL (Fiber Optic Inter Repeater Link) and 10 BASE-FB (Fiber Backbone), the maximum length of the segment is about 2000 meters.

[19] [19] M. Kemper Littman, Building Broadband Networks, “Figure 4.1,” chap. 4 in Building Broadband Networks, Auerbach Publications, 2002.

10BASE-5, 10BASE-2, and 10BASE-F installations employ bus or linear topologies. By comparison, 10BASE-T installations employ a star topology for linking network segments to a centralized hub.

Fast Ethernet networks are capable of transmitting at rates of 10 Mbps or 100 Mbps; Fast Ethernets can transmit across UTP or fiber optics. The standards developed are dependent on the type of cable used and are discussed below. Fast Ethernets are known as IEEE 802.3u standards.

The 100BaseTX network, also known as 100BaseT, is a standard for networking computers at 100 Mbps using Category 5 UTP cable. The workstation-to-hub or switch distance is limited to 100 meters. This standard uses the CSMA/CD method and can co-exist with 10BaseT networks. Fiber optics provide for greater cable lengths at 100 Mbps than UTP cable. With UTP, we are limited to two hubs between workstations, and the hubs must be connected using a 5-meter cable.

The 100BaseT4 network uses four-pair Category 3, Category 4, or Category 5 UTP cable. The 100BaseT4 operates at 100 Mbps, and the workstation-to-hub or switch distance is limited to 100 meters. This standard uses the CSMA/CD method and can co-exist with 10BaseT networks.

The 100BaseFX network is a variant of the 100BaseTX that uses multimode fiber-optic cable. It is normally used as a network backbone. The allowable distance between hubs is 400 meters.

The 1000BaseFX fiber optic network, more commonly known as Gigabit Ethernet, can transmit up to 1000 Mbps or 1 Gbps. Gigabit Ethernets are known as IEEE 802.3z standards. Figure 2.13 below shows a comparison between the various LAN technologies.

Standard	Topology	Cable Type	Data Rate (Mbps)	Max Number of Nodes per Segment	Min Distance Between Nodes (Meters)	Max Segment Length (Meters)
Ethernet						
10Base5	Bus	RG-8	10	100	2.5	500
10Base2	Bus	RG-58	10	30	0.5	185
10BaseT	Star	Cat 3–5 UTP	10	1024[a]	2.5[b]	100
10BaseFL	Star	Fiber Optic	10	1024	No minimum	2000
100BaseTX	Star	Cat 5 UTP	100	1024	2.5[c]	No maximum
100BaseT4	Star	Cat 3–5 UTP	100	1024	2.5	No maximum
100BaseFX	Star	Fiber Optic	100	1024	2.5	No maximum
1000BaseFX	Star	Fiber Optic	1000	1024	2.5	No maximum
Token Ring	Physical Star	UTP STP	4 16	UTP: 72 STP: 260	2.5	UTP: 45 STP: 101
	Logical Ring	Fiber Optic		33[d]		
FDDI	Ring	Fiber Optic	100	500	No minimum	200,000

a]Maximum number of network nodes and also maximum number of segments

[b]Maximum distance from node to hub

[c]Minimum distance between nodes

[d]Maximum number of segments

Figure 2.13 Characteristics of Ethernet Network types[20]

[20] Stephen T. Karris , "Table 4.3 lists the advantages and disadvantages of the four types of networks," chap. 4 in *Networks: Design and Management,* Orchard Publications, 2004.

2.2.5 Wide Area Network and Core Network technologies

Wide Area Network (WAN) is defined as a data (including voice and video) communication network which operates over a very large geographical area. It uses the transmission facilities provided by common or backbone data service providers such as the telephone companies.

WAN technologies generally work at the lower three layers of the OSI model: the physical layer, the data link layer and the network layer. Figure 2.14 below shows the OSI layers corresponding to the various WAN specifications as mentioned in [27].

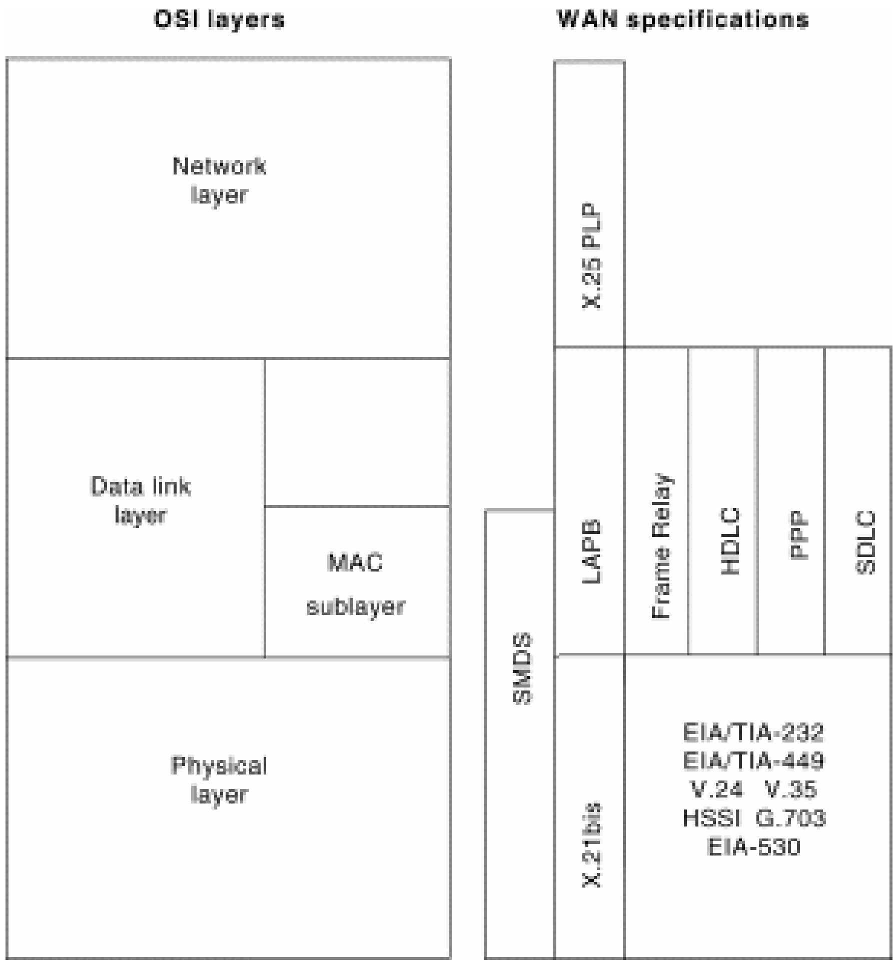

Figure 2.14 WAN Technologies and the OSI layers[21]

It uses both circuit switching and packet switching technologies. We have discussed circuit switching earlier; a few packet switching technologies are Asynchronous Transfer Mode (ATM), Frame Relay, Switched Multimegabit Data Services (SMDS), and X.25.

ATM was originally conceived for the purpose of high speed transport of video, voice and data over public networks. It is based on the specification provided by the ITU-T Broadband Integrated Services Digital Network (B-ISDN) standard. The ATM Forum extended the ITU-T's vision of ATM for use over public and private networks. ATM

[21] Cisco Documentation, "Figure 3-1: WAN Technologies Operate at the Lowest Levels of the OSI Model," chap. 3 in *Internetworking Technologies Handbook*, [online], Cisco Systems, Inc. http://www.cisco.com/univercd/cc/td/doc/cisintwk/ito_doc/introwan.htm

forum has recently merged with the MPLS and Frame Relay Alliance as per the press release [28] to form the MFA Forum. They have released the following specifications.

User-to-Network Interface (UNI) 2.0

- UNI 3.0
- UNI 3.1
- UNI 4.0
- Public-Network Node Interface (P-NNI)
- LAN Emulation (LANE)
- Multiprotocol over ATM

Frame Relay has been defined in [29] as, "Frame Relay is a high-performance WAN protocol that operates at the physical and data link layers of the OSI reference model. Frame Relay originally was designed for use across Integrated Services Digital Network (ISDN) interfaces. Today, it is used over a variety of other network interfaces as well. Frame Relay often is described as a streamlined version of X.25, offering fewer of the robust capabilities, such as windowing and retransmission of last data that are offered in X.25."

The Frame relay specifications were initially developed by a consortium formed by Cisco, Digital Equipment Corporation (DEC), Northern Telecom, and StrataCom in 1990. This consortium developed a specification that conformed to the basic Frame Relay protocol that was being discussed in CCITT (now ITU), but it extended the protocol with features that provide additional capabilities for complex internetworking environments. These Frame Relay extensions are referred to collectively as the Local Management

Interface (LMI). At present ITU-T provides international standards for frame relay, ANSI provides standards for the U.S.

MPLS (Multi Protocol Label Switching), an IETF initiative integrates Layer 2 information about network links (bandwidth, latency, utilization) into Layer 3 (IP) within a particular autonomous system or ISP in order to simplify and improve IP-packet exchange.

MPLS gives network operators a great deal of flexibility to divert and route traffic around link failures, congestion, and bottlenecks.

There are many other technologies which are used in the core network to provide high speed transport; we will discuss them as and when they are referred to in the following chapters. Figure 2.15 below gives an idea about these technologies working together to provide a converged network environment.

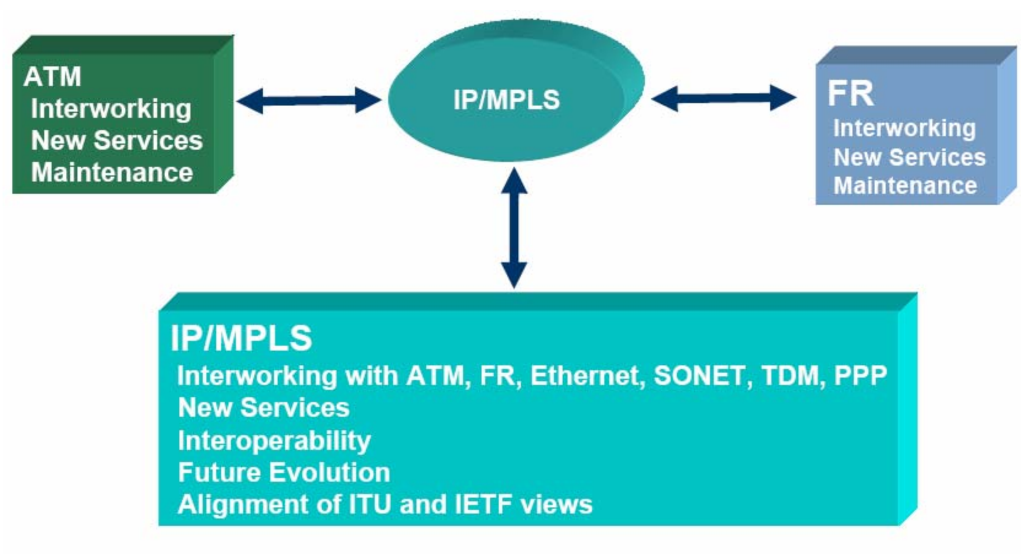

Figure 2.15 Packet Switching technology Convergence[22]

[22] MFA Forum, “Technology Overview,” MFA website, http://www.mfaforum.org/tech/index2.shtml

The internet is the largest WAN, which connects various networks and terminals throughout the world. It uses the TCP/IP (Transmission Control Protocol/ Internet Protocol) protocol suite as the common communication protocol. Figure 2.16 below shows a simplified architecture and operation of the internet environment. It can be seen that TCP operates at layer 4 of the OSI model. It performs the task of ensuring the successful and accurate transmission of the data. Internet Protocol on the other hand operates at layer 3 and provides a means of addressing and routing of information to the correct host.

IPv4 is being used at present, but it has certain shortcomings, in terms of number of possible address, security, and QoS features. The requirements for addressing schemes have become more demanding due to the rise in number of terminals and services being provided. IPv6 (Version 6) has been created to solve the above mentioned short comings. We will discuss more about this topic in the following chapters.

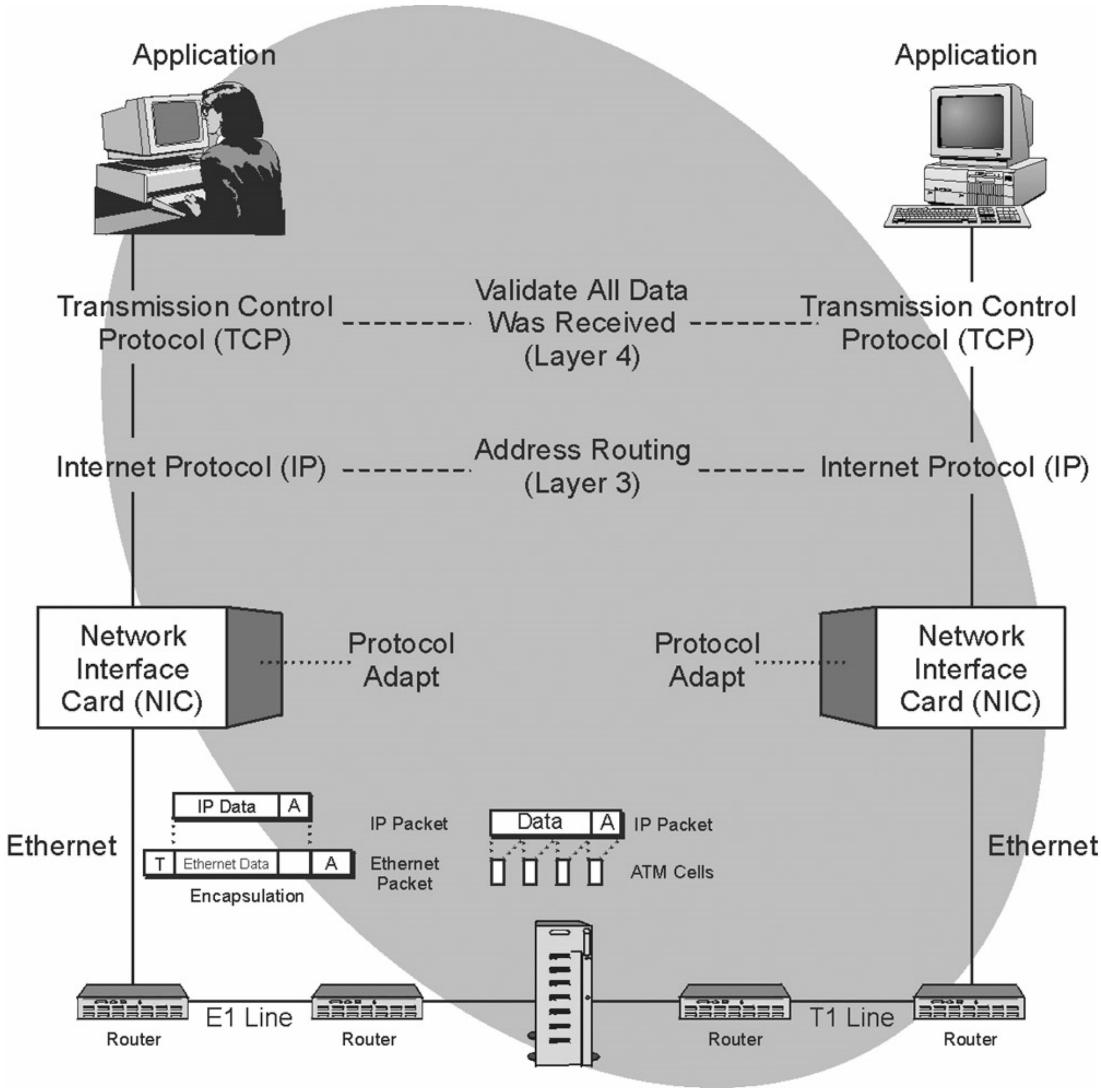

Figure 2.16 Simplified architecture of the internet[23]

2.3 Wireless Telecommunication systems

Wireless telecommunication is the main focus of the telecommunication industry at present. There are numerous advantages of wireless systems over wireline systems which have encouraged the rapid growth and continued interest in R&D. Wireless media has proven to be an effective and reliable alternate for traditional wireline communication.

[23] Lawrence Harte, W.E. Harrelson and Avi Ofrane , "Figure 9.17: Internet Data Routing" chap. 9 in *Telecom Made Simple*, APDG Publishing, 2002

The lower installation and operational costs are also a major factor. Wireless systems have gained popularity in personal mobile communication as well as in broadband internet access, especially in the LAN environment. Personal Area Network (PAN) technologies which operate over very small ranges, have shown great promise in terms of acceptance and usefulness.

2.3.1 Mobile personal communication systems

This section will provide information about the various generations of cellular communications to date and systems that are being developed for the future. It includes basic information about the technologies used to provide wireless services, their gradual improvement which enhanced the capacity of wireless communication, and the shift from pure voice, circuit switched environment to voice and data, packet switched systems. In this section attention is paid to the capacity (data communication) of the systems and not the technologies and standards used to attain them. As mentioned in the beginning the focus of this thesis is to investigate the convergence of various telecommunication systems at the application level, so the inherent technologies, though very important to provide the data communication, will not be discussed in detail.

2.3.1.1 Early system 1G and 2G

First generation cellular systems (1G) were designed exclusively to provide voice communication. The two main systems used worldwide were Advanced Mobile Phone System (AMPS) and Total Access Communication Services (TACS). AMPS was developed and introduced in North America in the early 1980's. It operates in the 800-MHz band (821 to 849 MHz) for base station receiving and (869 to 894 MHz) for base station transmitting. TACS was the European version of AMPS; the frequency range of operation is 890 MHz to 915 MHz for base receiving and 935 MHz to 960 MHz for base station transmitting. These systems did use digital signaling in many aspects of the network including the air interface but the content was transported in analog mode; in other words no CODEC was involved.

Figure 2.17 below provides a good comparison of the various 1G technologies in terms of attributes such as modulation, access method etc.

	AMPS	NAMPS	TACS	NMT450	NMT900
Base Tx MHz	869–894	869–894	935–960	463–468	935–960
Base Rx MHz	824–849	824–849	890–915	453–458	890–915
Multiple Access Method	FDMA	FDMA	FDMA	FDMA	FDMA
Modulation	FM	FM	FM	FM	FM
Radio Channel Spacing	30 kHz	10 kHz	25 kHz	25kHz	12.5kHz
Number Channels	832	2496	1000	200	1999
CODEC	NA	NA	NA	NA	NA
Spectrum Allocation	50MHz	50MHz	50MHz	10MHz	50MHz

Figure 2.17 1G technology platforms[24]

Narrow Band AMPS (NAMPS) was developed as an interim technology between 1G and 2G. It utilized 10 KHz voice channels and thus was able to provide three times the capacity in terms of available channels.

[24] C. Smith and D. Collins, "Table 2-1: 1G Technology Platforms," chap. 2 in *3G Wireless Networks*, McGraw-Hill, 2002.

Nordic Mobile Telephone (NMT) cellular standard was developed by the Nordic countries of Sweden, Denmark, Finland, and Norway in 1981. This type of system was designed to operate in the 450-MHz and in the 900-MHz frequency bands. All the systems used Frequency Modulation (FM) for modulating the voice signal and Frequency Division Multiple Access (FDMA) was the multiple access method.

The second generation of wireless telephony was a major advancement over the previous generation as it used digital radio technology to transport content. The reason for digitizing voice is to increase the efficiency of spectrum utilization; it allows better multiplexing techniques to be used. Also it allowed for improved voice quality as digital data was not subject to distortions in the way analog voice signals were. The aim was to take advantage of the various techniques and features that were not feasible for analog systems.

The migration from 1G to 2G involved the implementation of different digital techniques in the cellular area. The digital techniques for cellular communication fall into two primary categories: AMPS and the TACS spectrum. For markets employing the TACS spectrum allocation, the Global System for Mobile communications (GSM) is the preferred digital modulation technique. However, for AMPS markets, the choice is between Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA) radio access platforms. These services are referred to as Personal Communication Services (PCS) in the US. In addition to the AMPS/TACS spectrum, the IDEN radio access platform is available and it operates in the Specialized Mobile Radio

(SMR) band, which is neither cellular (GSM) nor PCS. Figure 2.18 below shows some technology platforms for 2G systems.

Cellular and SMR Bands	IS-136	IS-136*	IS-95	GSM	IDEN
Base Tx MHz	869–894	851–866	869–894	925–960	851–866
Base Rx MHz	824–849	806–821	869–894	880–915	806–821
Multiple Access Method	TDMA/FDMA	TDMA	CDMA/FDMA	TDMA/FDMA	TDMA
Modulation	Pi/4DPSK	Pi/4DPSK	QPSK	0.3 GMSK	16QAM
Radio Channel Spacing	30kHz	30kHz	1.25MHz	200kHz	25kHz
Users/Channel	3	3	64	8	3/6
Number Channels	832	600	9 (A) 10 (B)	124	600
CODEC	ACELP/VCELP	ACELP	CELP	RELP-LTP/ ACELP	
Spectrum Allocation	50MHz	30MHz	50MHz	50MHz	30MHz

Figure 2.18 2G technology platforms[25]

IS-136, also know as Digital AMPS (DAMPS) is used throughout the Americas. IS-136 defined in [31] adds a number of features to the original IS-54, such as text messaging, circuit switched data etc. It is a Time Division Multiple Access (TDMA) system.

Interim Standard 95 (IS-95) was the first CDMA (Code Division Multiple Access) based digital cellular standard pioneered by Qualcomm. The brand name for IS-95 is cdmaOne. It is a spread spectrum technology that allows multiple users to occupy the same radio channel. CDMA is based on the principal of Direct Sequence (DS) and is a wideband spread spectrum technology. The CDMA channel utilized is reused in every cell of the

[25] C. Smith and D. Collins, "Table 3-2: 2G Technology Platforms," chap. 3 in *3G Wireless Networks*, McGraw-Hill, 2002.

system and is differentiated by the pseudorandom number (PN) code that it utilizes. CDMA technology is used in the more recent systems such as WCDMA (UMTS) and CDMA 2000, EV-DO etc, which will be discussed later.

Global System for Mobile Architecture (GSM) was designed completely from scratch, unlike IS-136 and IS-95. The entire system was a new design including the air interface, network architecture, other interfaces and services. It did not provide backward compatibility with any existing analog system, as there were so many of them, especially in Europe. Figure 2.19 below shows the GSM system architecture.

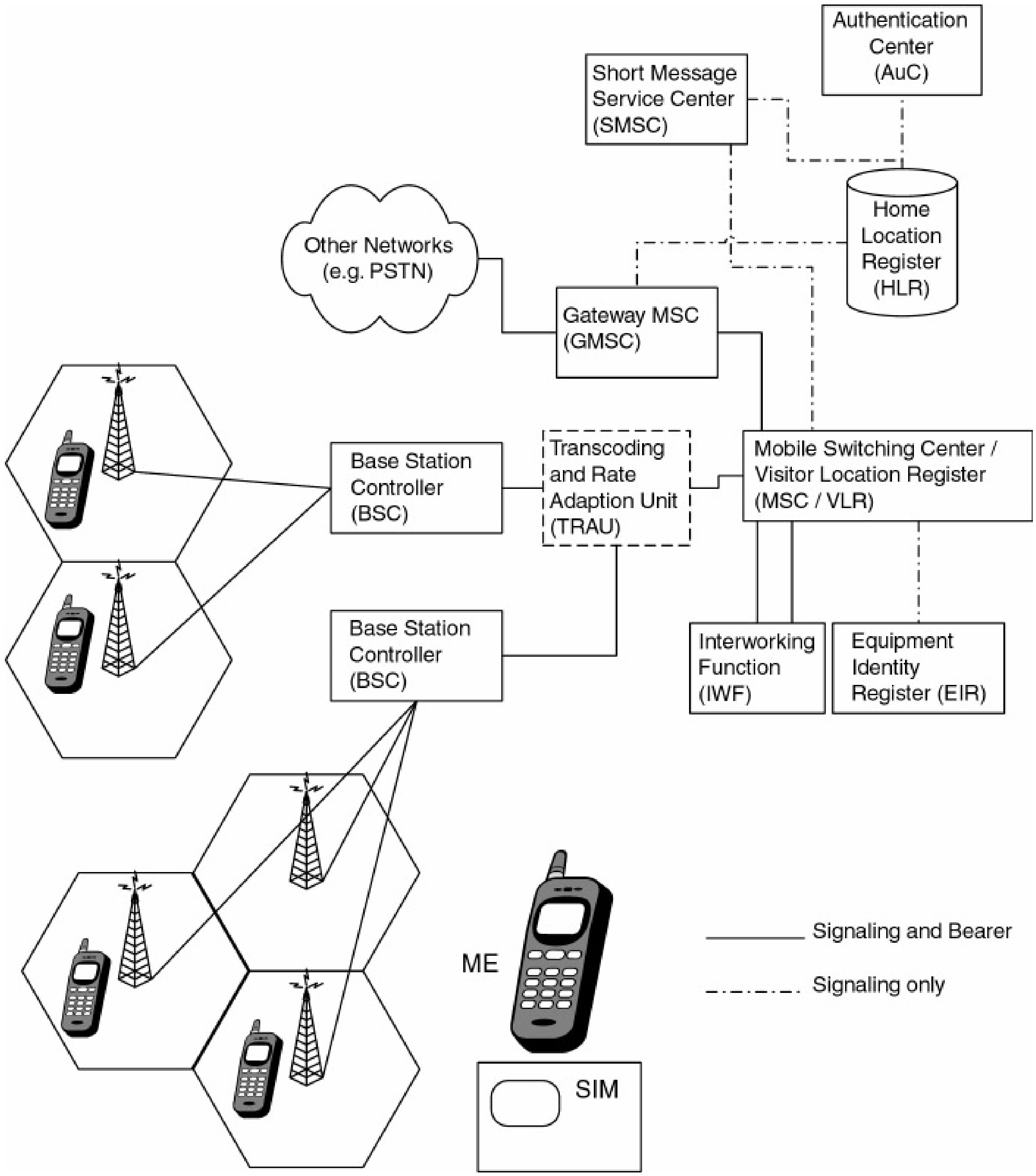

Figure 2.19GSM system architecture[26]

GSM is a TDMA system, with Frequency Division Duplex (FDD). The modulation scheme used is Gaussian Minimum Shift Keying (GMSK). GSM has been deployed in

[26] C. Smith and D. Collins, "Figure 3-6: GSM System Architecture," chap. 3 in *3G Wireless Networks*, McGraw-Hill, 2002

numerous frequency bands including the 900-MHz band, the 1800-MHz band, and the 1900-MHz band (in North America).

The IDEN (Integrated Dispatch Enhanced Network) system is a unique wireless access platform because it involves integrating several mobile phone technologies together and is based on a modified GSM platform. This system is both cellular and trunked in nature. It was developed by Motorola and is marketed in competition with both trunked and cellular radio. It is a digital narrow-band Time Division Multiple Access (TDMA) (squeezing 6 channels into 25 kHz) system, using a multilevel linear modulation scheme called M-16QAM. This particular modulation method has proven quite robust and has the ability to tolerate severe differential path delays.

Personal Digital Cellular (PDC) is another 2G cellular technology, developed and used exclusively in Japan. The standard was defined by ARIB (Association of Radio Industries and Businesses, formerly RCR) in 1991, and NTT DoCoMo launched its digital service in 1993. PDC uses 25 kHz carrier, 3 time slots, pi/4-DQPSK modulation and low bit-rate 11.2 Kbit/s and 5.6 Kbit/s (half-rate) voice codecs.

2.3.1.2 Cellular data communication: evolution from 2G - 2.5G

The rapid growth of the internet led to increased demand for data or IP services; it caused a shift in the mobile communication industry. Data services were being introduced along with voice services; this period of transition is referred to as 2.5G. During this period the service providers added packet based services to their existing voice networks. This packets based system did not necessarily provide faster services as the time slots or the existing spectrum was being used. A few of the technologies being used are mentioned below.

General Packet Radio Service (GPRS) is a very popular transition technology between 2G and 3G cellular networks. It is a mobile data service available for GSM users.

The GPRS systems was initially standardized by the ETSI (European Telecommunications Standards Institute) but is now being handled by the 3GPP (3^{rd} Generation Partnership Project). The GPRS network architecture reuses the GSM network nodes such as MSC/VLR, HLR, and BSS (shown in figure 2.20). New network nodes have been introduced for the transport of packet data. These nodes are the Gateway GPRS Support Nodes (GGSN) and Serving GPRS Support Nodes (SGSN). The subnetwork formed by the SGSNs and the GGSNs is called the GPRS core network. GPRS provides data rates up to 128 Kbps and can handle protocols such as IP, PPP(Point to Point Protocol) and X.25 to provide various data services.

Enhanced Data rates for GSM Evolution (EDGE) is an enhanced version of GPRS and works on TDMA and GSM networks. It provides data rates of up to 384 Kbps and can be used for any packet switched application such as internet access. EDGE is the final evolution of data communications within the GSM and IS-136 standards. EDGE uses 8PSK modulation at high data rates and standard GMSK at lower data rates.

High-Speed Circuit-Switched Data (HSCSD) is another 2.5G technology to provide high bit rate data communication on existing GSM systems. It can provide data rates of 28.8/ 56 Kbps.

CDMA2000 is a 3G mobile telecommunication standard, and is a registered trademark of (TIA-USA). CDMA2000-1xRTT (Radio Transmission Technology), is the basic layer of CDMA2000, it supports data rates of up to 144 Kbps, and requires 1.25 MHz spectrum for its operation. IS-2000 is a second generation CDMA system and an extension of IS-95. It is an early form of CDMA2000.

Due to the high bit rates of EDGE and CDMA2000-1xRTT they are sometimes classified under 3G technologies, the definition of 3G system is given in the following section.

Figure 2.20 below gives a side by side comparison of different 2.5G technologies.

2.5G Technology	Data Capability	Spectrum Required	Comment
HSCSD	28.8/56 Kbps	200 kHz	Circuit/Packet Data
GPRS	128 Kbps	200 kHz	Circuit/Packet Data
Edge	384 Kbps	200 kHz	Circuit/Packet Data
CDMA2000-1XRTT	144 Kbps	1.25 MHz	Circuit/Packet Data

Figure 2.20 Technologies compared 2.5G[27]

2.3.1.3 3G and beyond 3G systems

Third Generation wireless communication (3G) systems were originally been defined by the ITU-R recommendation ITU-R M.1457 [32]. It has been superseded; the latest document is [33]. It is a radio and network access specification defining several methods or technology platforms that meet the overall goals of the specification. The IMT-2000

[27] C. Smith and D. Collins, “Table 4-2: 2.5G and 3G Comparison,” chap. 4 in *3G Wireless Networks*, McGraw-Hill, 2002

specification is meant to be a unifying specification, enabling mobile and some fixed high-speed data services to use one or several radio channels with fixed network platforms for delivering the following envisioned services as mentioned in [30].

- Global standard
- Compatibility of service within IMT-2000 and other fixed networks
- High quality
- Worldwide common frequency band
- Small terminals for worldwide use
- Worldwide roaming capability
- Multimedia application services and terminals
- Improved spectrum efficiency
- Flexibility for evolution to the next generation of wireless systems
- High-speed packet data rates
    - 2 Mbps for fixed environment
    - 384 Mbps for pedestrian
    - 144 Kbps for vehicular traffic

Figure 2.21 below shows the linkage between various platforms that comprise IMT-2000 specification group.

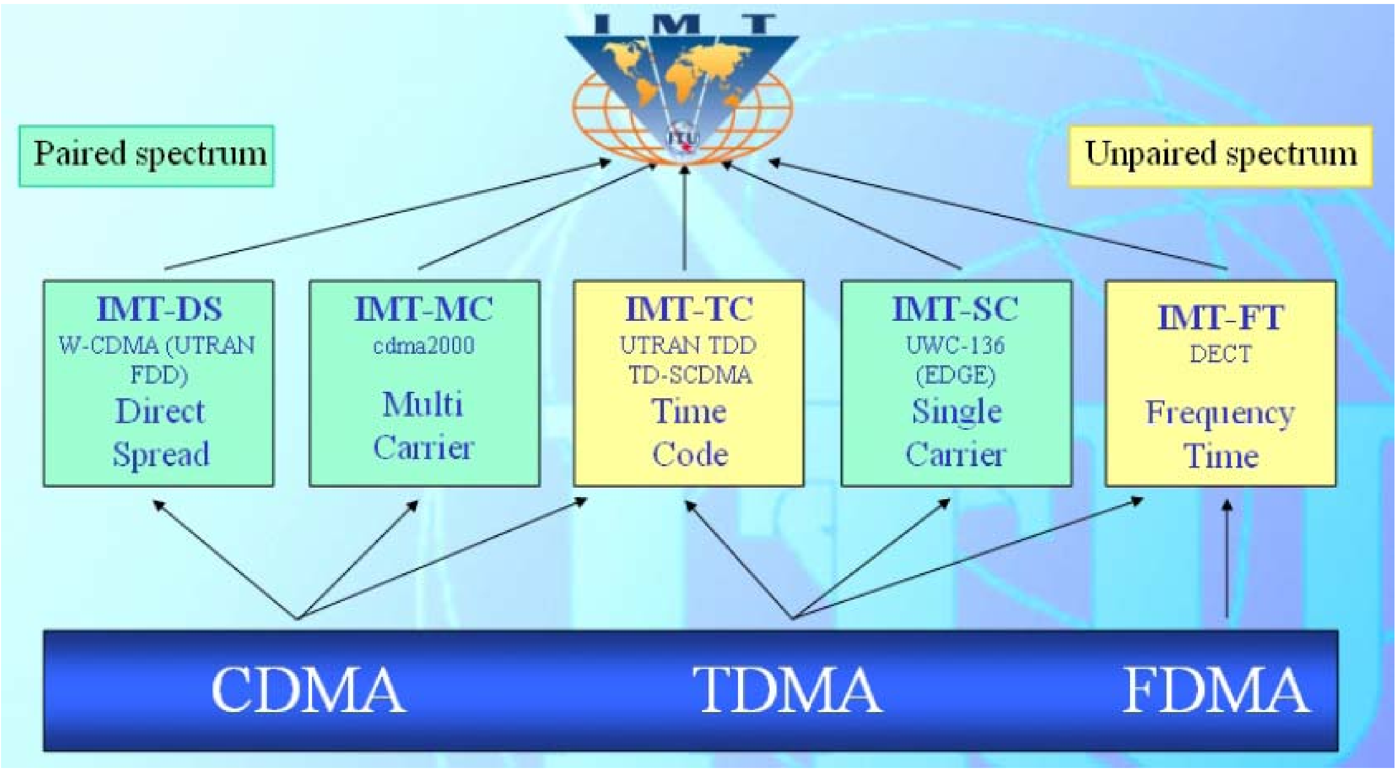

Figure 2.21 Radio platforms included in IMT-2000 specification[28]

The scope of IMT-2000 specifications is very broad; the purpose of including various radio technology options is to allow for a seamless service evolution from the various 2G mobile standards; that are extensively deployed around the world.

The migration path as suggested by IMT is shown below in figure 2.22.

[28] ITU, "IMT 2000 Project," ITU website, http://www.itu.int/osg/imt-project/docs/What_is_IMT2000-2.pdf

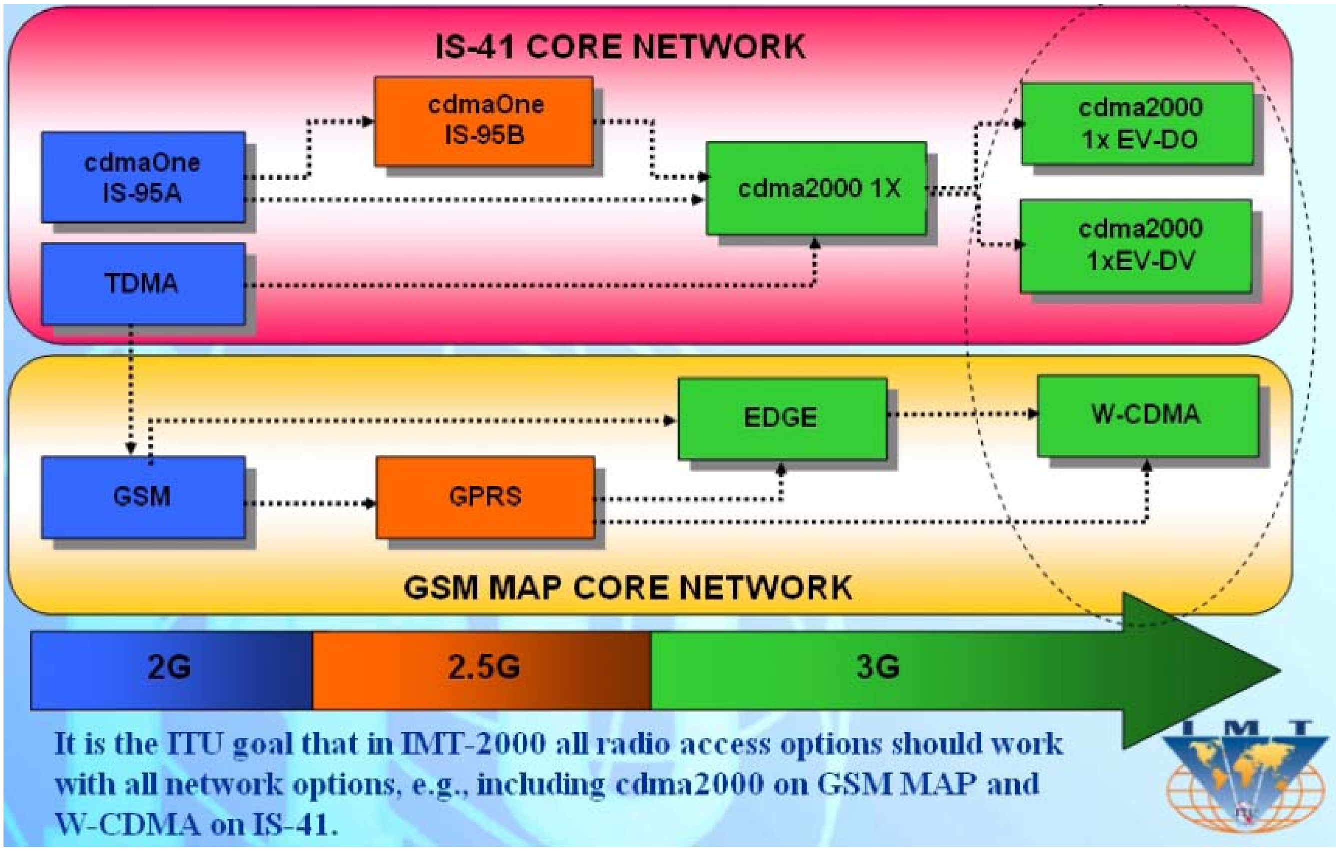

Figure 2.22 Migration paths for mobile communication systems[29]

As mentioned earlier, IMT-2000 included five different air interfaces. Different standardization agencies agreed on different air interfaces. ETSI (European Telecommunication Standards Institute) accepted a WCDMA (Wideband CDMA) approach using FDD (Frequency Division Duplex). In Japan a WDCMA solution was proposed with both FDD and TDD. In North America a CDMA based solution was proposed, which was an extension of IS-95 (CDMA2000).

It was clear that a number of different organizations were to work on similar technologies, so a decision was made to pool the resources and two organizations were formed, the 3rd Generation Partnership Project (3GPP) and the 3rd Generation

[29] ITU, "IMT 2000 Project," ITU website, http://www.itu.int/osg/imt-project/docs/What_is_IMT2000-2.pdf

Partnership Project 2 (3GPP2). 3GPP has defined a mobile system that fulfills the IMT-2000 standard. It is called Universal Mobile Telecommunication System (UMTS) and is based on WCDMA. 3GPP2 has defined the CDMA2000 standard which builds on the 2G CDMAone standard.

The purpose and organization of 3GPP as mentioned in [34] is given below.

> 3GPP is a collaboration agreement that was established in December 1998, the collaboration agreement brings together a number of telecommunications standards bodies which are known as "Organizational Partners". The current Organizational Partners are ARIB (Association of Radio Industries and Businesses), CCSA (China Communications Standards Association), ETSI (European Telecommunication Standards Institute), ATIS (Alliance for Telecommunications Industry Solutions), TTA (Telecommunications Technology Association), and TTC (Telecommunications Technology Committee).
>
> The original scope of 3GPP was to produce globally applicable Technical Specifications and Technical Reports for a 3rd Generation Mobile System based on evolved GSM core networks and the radio access technologies that they support (i.e., Universal Terrestrial Radio Access (UTRA) both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) modes). The scope was subsequently amended to include the maintenance and development of the Global System for Mobile communication (GSM) Technical Specifications and Technical Reports including evolved radio access technologies (e.g. General Packet Radio Service (GPRS) and Enhanced Data rates for GSM Evolution (EDGE)).

> The discussions that led to the signing of the 3GPP Agreement were recorded in a series of slides called the "Partnership Project Description" that describes the basic principles and ideas on which the project is based. The Partnership Project Description has not been maintained since its first creation but the principles of operation of the project still remain valid.

3GPP provides specifications in the form of releases. They were done annually, and all the releases prior to 1999 were exclusively for GSM and its derivatives. Release '99 was the first release to provide basic standards for 3G services. This was followed by Release 4, Release 5, Release 6 and recently Release 7. Each release advanced the specifications that were provided in the previous release. Information about the process of developing releases and the groups involved will the mentioned in the following chapter when we discuss IMS (Internet Protocol Multimedia Subsystem), which was introduced in Release 5.

3GPP2 is the other collaborative effort that was also launched in 1998. It comprises of the five organizational partners, ARIB (Japan), TIA (Telecommunication Industry Association) (USA), TTA (Korea), TTC (Japan) and CCSA (China). Besides these Standard Development Organizations (SDO), 3GPP2 comprises of Market Representation Partners (MRP) who offer market advice to 3GPP2 and bring a consensus view of market requirements (e.g., services, features and functionality) falling within the 3GPP2 scope. The MRPs include the CDMA Development Group (CDG), IPv6 Forum, and International 450 Association (IA 450).

The 3GPP2 specifications are produced by the project's four Technical Specification Groups (TSG). The TGSs are TSG-A (Access Network Interfaces), TSG-C (cdma2000®)

TSG-S (Services and Systems Aspects), and TSG-X (Core Networks). Additional information about 3GPP2, its operation and specifications can be obtained from their website [35]. 3GPP2 has produced specifications for systems such as CDMA2000 1xEV-DO (data optimized) and 1xEV-DV (data and video). The evolution path of the CDMA2000 is shown below in figure 2.23

Figure 2.23 Evolution of CDMA2000 1x[30]

The organizations mentioned above are among the major players involved in the development of next generation or beyond systems. They are organized and have a certain mechanism in place to plan, experiment and introduce new technology standards, which would be globally applicable.

Efforts have already begun to develop beyond 3G systems, this includes 3.5 G systems such as HSDPA (High Speed Downlink Packet Access) and HSUPA (High Speed Uplink Packet Access), developed by 3GPP, and presented in Release 6. This technology is defined in [36] as following, “HSDPA is a packet-based data service in W-CDMA downlink with data transmission up to 8-10 Mbps (and 20 Mbps for MIMO systems) over a 5MHz bandwidth in WCDMA downlink. HSDPA implementations include Adaptive Modulation and Coding (AMC), Multiple-Input Multiple-Output (MIMO), Hybrid Automatic Request (HARQ), fast cell search, and advanced receiver design.”

[30] C. Smith and D. Collins, “Figure 4-7 CDMA2000-1X evolution process,” chap. 4 in *3G Wireless Networks*, McGraw-Hill, 2002

The above mentioned technologies such as MIMO, AMC, HARQ, etc along with the rapid improvement in the hardware technology and improved air interface mechanisms such as OFDMA (Orthogonal Frequency Division Multiple Access) have proven to be major enabling technologies for the development of high bit rate wireless communication systems.

2.3.2 Wireless Local Area Networks (WLAN)

The local area network has been around for a long time; it's defined by the IEEE 802.3 standard. WLAN uses radio waves as its carrier and it provides Ethernet connectivity to the user without the restrictions of wires. Various standards are available for WLANs which provide data connectivity at different rates and using different frequencies.

The IEEE 802.11 committee was set up to define specifications and functional standards for wireless LANs. The vast majority of wireless LAN systems in use today conform to an IEEE 802.11 specification [39]. A common medium access control layer is defined across a number of different physical layers, including infrared.

A few IEEE WLAN standards with their range and data rates are mentioned below

- 802.11a/h – 50m, 54 Mbps
- 802.11b – 100m, 11Mbps
- 802.11g – 100m, 54Mbps
- 802.11i – overlay with authentication and encryption
- 802.11c - to improve interoperability between devices

- 802.11d - to improve roaming
- 802.11e - for improved quality of service
- 802.11f - to regulate inter-access point hand offs
- 802.16e – WiMax (based on 802.16a), for stationary and slow moving mobile end points

  30 mile radius

  Up to 70Mbps
- 802.20 – WMAN for end points at high speeds (up to 155 mph)

Some information about the more popular of the above mentioned standards is provided below.

The 802.11b [40] was released in 1999. It operates in the 2.4 GHz band and has a maximum raw data rate of 11 Mbit/s and uses CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) media access method defined in the original standard. Due to the CSMA/CA protocol overhead, in practice the maximum 802.11b throughput that an application can achieve is about 5.9 Mbit/s over TCP and 7.1 Mbit/s over UDP.

The 802.11a standard uses the same core protocol as the original standard, operates in 5 GHz band, and uses a 52-subcarrier Orthogonal Frequency-Division Multiplexing (OFDM) with a maximum raw data rate of 54 Mbit/s, which yields realistic net achievable throughput in the mid-20 Mbit/s.

802.11g was released in June of 2003. It works in the 2.4 GHz band (like 802.11b) but operates at a maximum raw data rate of 54 Mbit/s, or about 24.7 Mbit/s net throughput like 802.11a. It is fully backwards compatible with 802.11b and uses the same frequencies.

All these systems operate in the unregulated or unlicensed spectrum; the 2.4 GHz ISM band covers 2.400 GHz to 2.500 GHz and is designated worldwide for license-exempt industrial, scientific and medical (ISM) use. The 5 GHz radio LAN bands, 5.150–5.250 GHz, 5.250–5.350 GHz, and 5.470–5.725 GHz have been identified in most parts of the world for use to provide WLAN services; details are provided in [41].

The IEEE 802.16 Working Group has developed point-to-multipoint broadband wireless access standard for systems in the frequency ranges 10-66 GHz and sub 11 GHz. The standard covers both the Media Access Control (MAC) and the physical (PHY) layers as mentioned in [42]. The current 802.16 standard is IEEE Std. 802.16-2004 [45], approved in June 2004. IEEE Std. 802.16-2004 addresses only fixed systems. An amendment 802.16e is being developed by IEEE 802.16 Working Group; it adds mobility components to the standard. This amendment is expected to be introduced by end of 2005. Figure 2.24 below shows the utility of WiMAX.

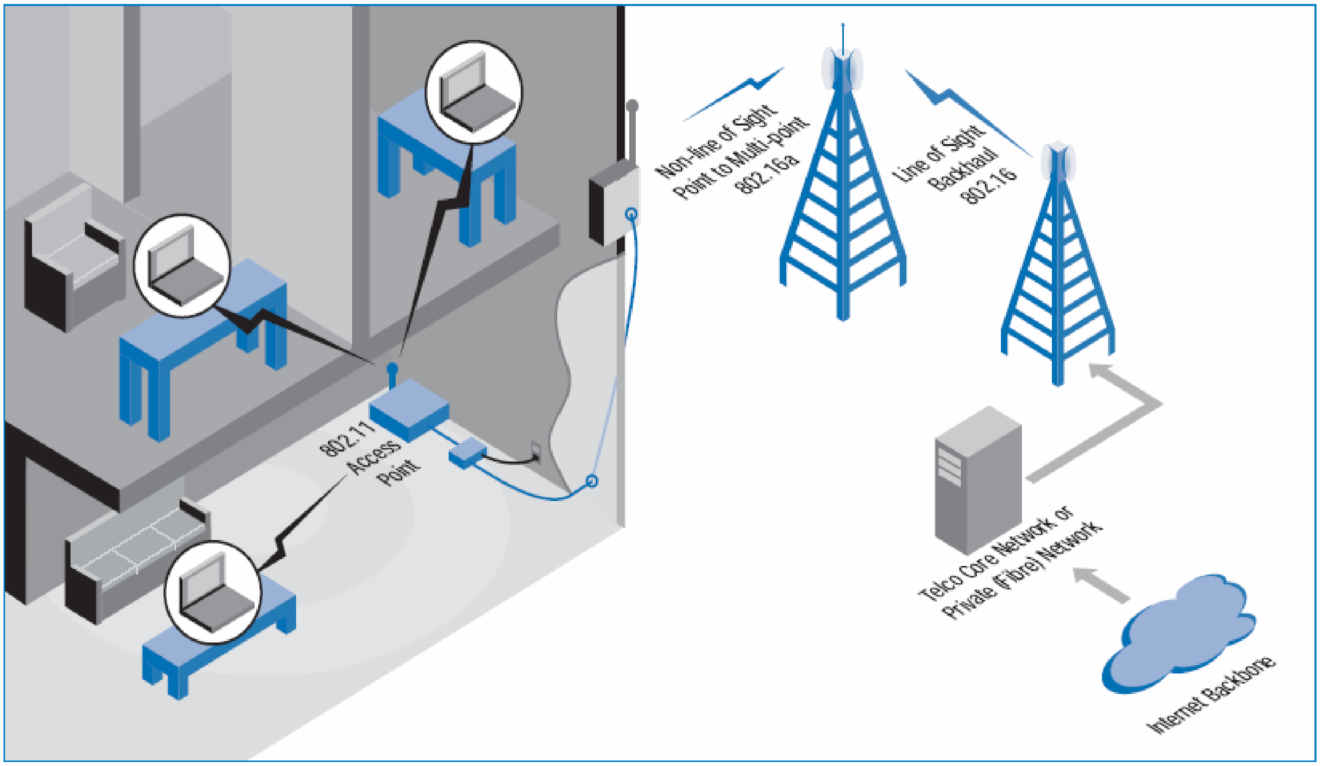

Figure 2.24 Broadband Wireless Access network using IEEE 802.16[31]

WLANs can operate in different network architectures, such as point to point, point to multipoint, ad hoc, etc; its operation will be discussed in more depth in the following chapters.

2.3.3 Wireless Personal Area Networks (WPAN)

A personal Area Network can be defined as a small network consisting of various devices (including computers, printers, telephone, fax, etc) close to a person. PANs can be used

[31] Intel White Paper, *IEEE 802.16* and WiMAX, Broadband Wireless Access for Everyone*, July 2003.

for communication among the personal devices themselves (intrapersonal communication), or for connecting to a higher level network and the internet.

A Wireless PAN can be established by using short range wireless technologies such as Bluetooth, IR (Infra Red) and Zigbee etc. PAN networks are defined by IEEE 802.15 standards. In March 1998, the Wireless Personal Area Network (WPAN) study group was formed.

In May 1998, the Bluetooth Special Interest Group (SIG) [44], Inc. was formed, and in May 1999 the IEEE WPAN Study Group became IEEE 802.15, the WPAN Working Group. In July 1999, Bluetooth released the Bluetooth Specification v1.0a. The 802.15.1 standards work is a cooperative effort with the Bluetooth SIG, Inc. This cooperative effort resulted from a convergence of IEEE standards development activities underway coupled with the formation of the Bluetooth SIG in 1998. IEEE Std 802.15.1™-2002 [43] was published 14 June 2002.

The Bluetooth vision is to cut the cord between devices, providing short range radio links to interconnect equipment. The idea is to provide Bluetooth functionality in all consumer and business equipment, eventually as a built-in capability rather than an add-on accessory. Transmission is in the 2.4 GHz ISM band using GFSK modulation and frequency hopping spread spectrum at 1600 hops per second with a 1 Mbit/s signalling rate [41].

2.3.4 4G- Next Generation Wireless Systems

The fourth generation wireless communication systems are expected to provide higher data rates of 100 Mbps and possibly up to 1 Gbps. Successful trial have been performed at Gigabit rates by NTTDoCoMo, and reported in [37].

There are various visions of the next generation systems, various technologies are being tried out, but certain aspects of the system are widely accepted. It is being designed to be all IP, system with horizontal and vertical handoff capabilities. This allows for a backward compatible system; such a vision is presented in [38]. Figure 2.25 below shows the nature of the system.

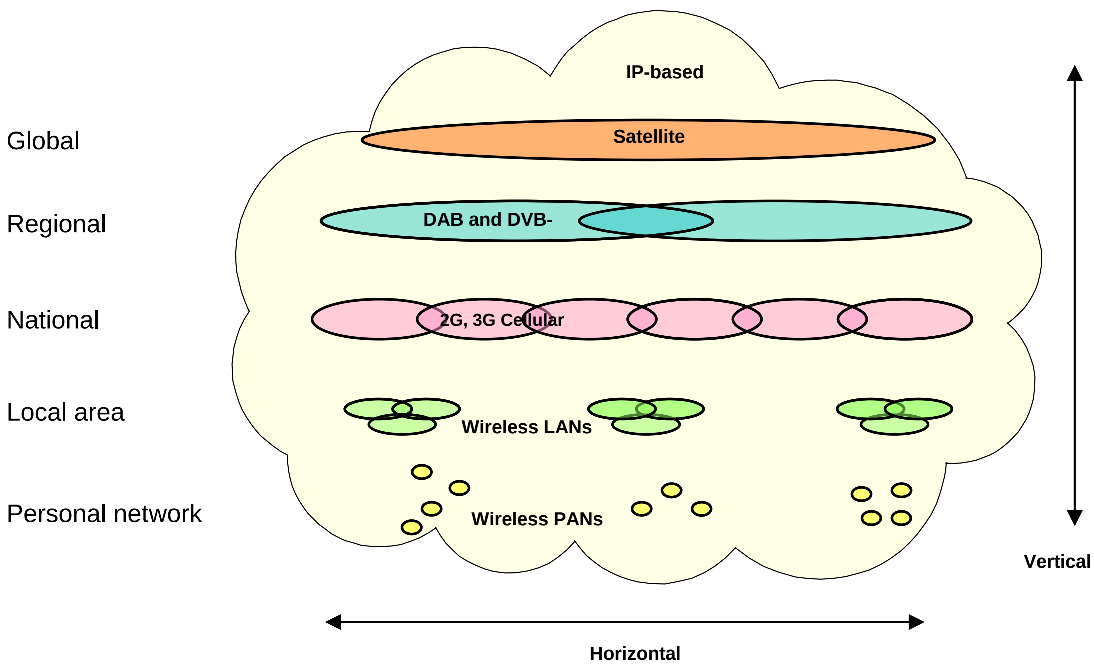

Figure 2.25 Hierarchical layers for 4G [32]

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[32] Prof. Hamid Aghvami, "A Vision for 4G" [online], Centre for Telecommunications Research - King's College London, Wireless Multimedia Communications Ltd, Dec 2001. http://www.iee.org/OnComms/pn/multimediacomms/Aghvami.pdf

There are various hurdles in the path of 4G systems; they are not just technical but include many other aspects such as market forces or customer demand, policy and political factors as well as security and reliability issues. Some of these cannot be predicted or controlled but efforts are being made internationally to solve technical issues and provide uniform approach towards the next generation.

2.4 Conclusions

This chapter provided a picture of the wireless and wireline telecommunication systems being used at present and to be introduced in the near future. It can be seen that all the systems that are being introduced are moving towards an all IP environment, and at the same time efforts are being taken to standardize the systems globally. All this supports the development of a converged and internetworked communication system that would allow for features like global roaming, single profile, and universal access to high data rate systems for advanced services.

In the following chapter we will look at the steps that are being taken to develop such a converged network; it has been referred to as the Next Generation Network (NGN). We will examine the need for NGN and the major driving forces; also we will look at the requirements of such a system, the organizations and procedures involved in developing it.

3. The push towards next generation systems

3.1 Introduction

The next generation networks (NGN) or advanced networks and systems are being developed in different parts of the world and by various agencies and organizations. Most of them are working in tandem to develop a uniform standardized system. The purpose of this chapter is to clearly identify the reasons for these efforts and their objectives.

This chapter would attempt to define the requirements of these future networks in terms of capacity, security, management and other such aspects of data communication. Also we will try and identify the main drivers or reasons for these systems being given so much importance.

There are a number of different visions for the next generation of communication systems; most of them attempt to provide similar data communication services and abilities to the end users, though the way in which they attempt to achieve these outcomes may be different.

We will examine the requirements of the NGN systems as defined by ITU and will look at the various organizations working together to obtain a standardized solution for the same. As mentioned earlier there are a number of agencies working towards the goal of advanced of future networks; a major contributor to this effort is 3GPP and 3GPP2, who introduced the IMS, which is the basis of work being done by many others.

In the United States, ATIS (Alliance for Telecommunication Industry Solutions) is leading the way in developing NGN systems based on extended IMS. We will look into their operations in some depth.

3.2 Next Generation Networks (NGN): Definition

To date there is not a single precise definition for the Next Generation Networks. It has been modified often; the most current definition as provided by ITU is as below [46].

> A Next Generation Network (NGN) is a packet-based network able to provide services including Telecommunication Services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users.

The NGN is characterized by the following fundamental aspects:

- Packet-based transfer
- Separation of control functions among bearer capabilities, call/session, and application/ service
- Decoupling of service provision from network, and provision of open interfaces
- Support for a wide range of services, applications and mechanisms based on service building blocks (including real time/ streaming/ non-real time services and multi-media)
- Broadband capabilities with end-to-end QoS and transparency
- Interworking with legacy networks via open interfaces
- Generalized mobility
- Unrestricted access by users to different service providers

- A variety of identification schemes which can be resolved to IP addresses for the purposes of routing in IP networks
- Unified service characteristics for the same service as perceived by the user
- Converged services between Fixed/Mobile
- Independence of service-related functions from underlying transport technologies
- Compliant with all regulatory requirements, for example concerning emergency communications and security/privacy, etc.

All the work being done towards the development of NGN systems is in conformity to the above mentioned fundamental characteristics. We will examine the work done so far by ATIS (Alliance for Telecommunications Industry Solution), and more specifically by the ATIS Next Generation Network – Focus Group (NGN-FG), which is commissioned by the ATIS Technology and Operations Council (TOPS). They have prepared and published a couple of documents which clarify their target objectives and features of the NGN being developed. The first document was released in November 2004 and a part of the executive summary given below clearly presents the purpose of the release [47].

> *The ATIS NGN Framework Part I document contains a snapshot of NGN target objectives and features, for which phased implementation requirements will be developed. The NGN-FG will continue to clarify the priorities (i.e.,, short-term, medium-term, long-term) for these standards initiatives.*
>
> *A key motivation for the NGN is to focus on the variety of new, value-added, IP-centric services and applications. An equally important motivation is to reduce Capital Expense (CAPEX) and Operational Expense (OPEX) through more efficient utilization of network resources to provide services (e.g. voice services).*

The NGN architecture builds on the ETSI TISPAN extended IP Multimedia System (IMS) session-based architecture to consistently support new value-added services. The ATIS NGN architecture may be further enhanced or modified to support other services provided by NGN Service Providers (e.g. broadband services, L2VPN, L3VPN). The PSTN Emulation subsystem is identified as a mechanism to facilitate migration from legacy PSTN services to NGN. This enables expense reduction through more efficient voice services, while still allowing the NGN to be optimized for the future SIP services model. The document identifies key IP access network requirements for compatibility with the core NGN. These include providing IP connectivity, QoS, and policy enforcement.

Figure 3.1 below provides a good idea about the intended converged NGN system. It comprises of traditional and advanced wireless and wireline services, including both voice and data services on either medium. It also comprises of the customer premises equipment (local networks etc). The common area is the focus of converged user centric services, when deployed these services will no longer be associated with the type of network access, but rather with the user need and services subscribed.

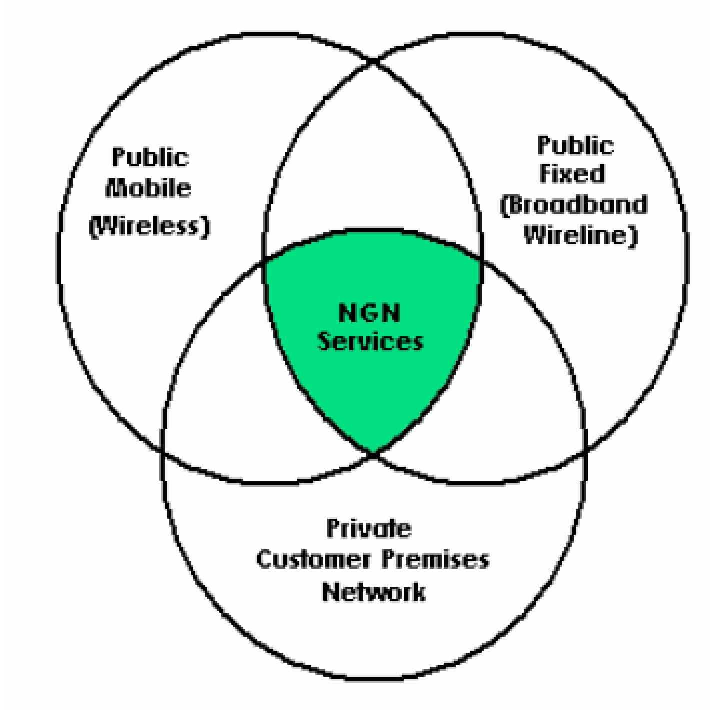

Figure 3.1 NGN Context: Convergence of Wireline, Wireless and Customer equipment[33]

3.3 ATIS: NGN Definitions, Requirements and Architecture

ATIS NGN-FG is driven by the business needs of the North American market; the aim is to produce as much as practically possible international NGN standards. For this reason it works in collaboration with a number of global standardization agencies.

Figure 3.2 below shows the interaction of ATIS with other organizations and agencies involved in the standardization process of next generation systems.

[33] ATIS, "Figure 1 - NGN Context: Convergence of Wireless, Wireline, and Customer Premise Services," in *ATIS Next Generation Network (NGN) Framework, Part I: NGN Definitions, Requirements, and Architecture, Issue 1.0,* November 2004, pg 10

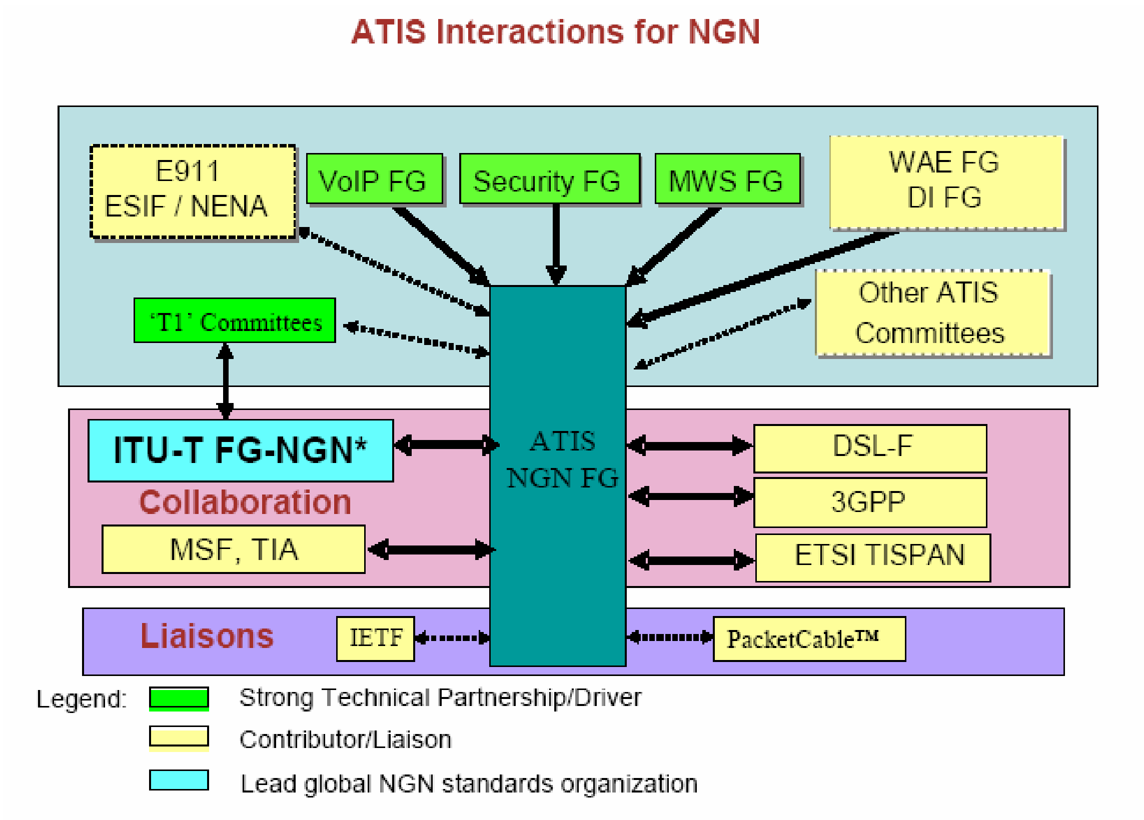

Figure 3.2 ATIS Standards Collaboration [34]

A list of the partners shown in the figure above is given below.

- ITU-T: SG13, including the Focus Group on Next Generation Networks (FGNGN)
- 3rd Generation Partnership Project (3GPP)
- European Telecommunications Standards Institute (ETSI) TISPAN
- Multiservice Switching Forum (MSF)
- DSL Forum (DSL-F)

[34] ATIS, "Figure 2 - Potential Standards Collaboration," in *ATIS Next Generation Network (NGN) Framework, Part I: NGN Definitions, Requirements, and Architecture, Issue 1.0,* November 2004, pg 11

- CableLabs™
- Institute for Electrical and Electronics Engineers (IEEE)
- ATIS Technical Committees *(e.g.*, PTSC, TMOC, NIPP)
- 3rd Generation Partnership Project #2 (3GPP2)
- Telecommunications Industry Association (TIA)
- Internet Engineering Task Force (IETF)
- National Emergency Numbering Association (NENA)
- Emergency Services Interconnection Forum (ESIF)
- TTY Forum
- Industry Numbering Committee (INC)
- FCC Network Reliability & Interoperability Council (NRIC)
- Open Mobile Alliance (OMA)
- Metro Ethernet Forum (MEF)
- MPLS and Frame Relay Alliance

ITU-T study groups including SG-13 and its Focus Group on Next Generation Networks (FGNGN) are responsible for the global standards for NGN telecommunication systems. The idea is to expedite the process of standards development by having regional Standard Development Organization (SDO) submit contributions to the ITU-T. Such a workflow provides and establishes a global scope to the efforts of ITU-T and allows for a more harmonized approach.

TISPAN is the ETSI group responsible for all aspects of standardization for present and future converged networks, including the NGN and including service aspects, architectural aspects, protocol aspects, QoS studies, security related studies, and mobility aspects within fixed networks, using existing and emerging technologies. TISPAN is developing an architecture for NGN based on the 3GPP IMS subsystem (release 6). This architecture is named "extended IMS" throughout this document and is considered the base for ATIS NGN work. More information about IMS will be provided in the following Chapter.

We have examined the role played by 3GPP and 3GPP2 earlier in chapter 2; both these organizations are closely aligned with the ITU-T 3G standards and play major roles in the development of NGN systems.

There are many other organizations and agencies which ATIS partners with to achieve their goals and objectives. Different agencies define standards for different mediums of communication, so it is essential for them to interact and develop interoperable standards. For example, DSL Forum, defines standards for data communication over the telephone lines, CableLabs® develops standards for communication over the cable infrastructure and IEEE has been developing and defining standards for the data communication for the LAN/MAN, in both wired (IEEE 802.3) and wireless Medium (IEEE 802.11 -WiFi and 802.16 - WiMax)

3.3.1 ATIS: NGN Requirements

The ATIS NGN Framework documents Parts I and II [47, 49] provide a detailed set of requirements for the NGN systems. These documents examine every aspect of telecommunications and clearly define the expectations from the NGN systems being developed. They are referred to as High Level requirements/ Guiding Principles for the system.

They have been divided into divided into six major groups of requirements

- General Requirements
- US Regulatory Requirements
- End User Applications
- Network Service Enablers
- Underlying Network / Support Capabilities
- Business Model Driven Requirements

Detailed information will not be provided about each of these categories, but a general idea about the contents of each of them is given below.

The general requirements section of the document contains guidelines about the basic functionality that would need to exist for the NGN to fulfill the ITU-T requirements.

It includes issues such as NGN network interconnection requirements, different types of interfaces between the Application Service Providers (ASP) and the Next Generation Service Providers (NGSP), mechanisms to measure and predict QoS, guidelines for incremental replacement of legacy services, PSTN Simulation, PSTN Emulation, mobile network evolution, transparent end to end communication, synchronization and timing issues, etc. This section of the document [47] provides all the necessary general guidelines for developing NGN networks.

The U.S. regulatory requirements section of the document lays down all the possible regulatory requirements that might be enforced on the NGN systems; these have been introduced and enforced on the traditional voice communication systems. The regulatory requirements are a moving target and may evolve with the regulatory and legislative actions of the various government agencies at different points of time. These requirements include information and guidelines regarding various service specific or general regulations such as Lawfully Authorized Electronic Surveillance (LAES), Number portability regulations, number pooling, E 9-1-1 directives, Emergency Telecoms Service (ETS), FCC rules and regulation, accounting etc.

The End User Applications section of the document provides guidelines regarding the user applications and about the way they would need to be approached in the new NGN network architecture. The NGN would require application to be supported on a common, converged architecture so there would be need for changes or variations in the services and their inherent capabilities. This section lays down requirements for applications such as interactive voice, content and capabilities of video services, multimedia conferencing, content sharing, basic and advanced interactive gaming, sensor and control networking, mobility management, etc.

Some of the key Network Service Enablers are defined in the next section of the document; even though the NGN is not vertically integrated, there are still some network based communication services that bring value to customer applications. These include

Quality of Service requirements, Presence services, Policy definition and enforcement, Media resource and Media Gateway Functions, Personal Profiles Unified Interface and Service Ubiquity. There are other important aspects which have been defined and described in this document; roaming, location services, personal information management and access, digital rights management and sessions management are some of them.

The underlying network/ support capabilities are those capabilities that are not directly accessible by the applications. These are of monumental importance and need to be introduced in the core network and are very well defined in this section of the document. These capabilities include, Operations Administration Maintenance and Provisioning (OAMP), security provision, integrity requirements, confidentiality and privacy requirements, attack mitigation and prevention policies, accounting (Ordering and Billing), trust policies and requirements, etc.

The last segment of the requirements identifies and defines the various Business Model Driven Requirements. This section covers topics such as the operational expense (OPEX), implications for service providers, third party access implications, service delivery environment, and consolidated operations requirements.

3.3.2 ATIS NGN Converged Architecture

Based on the NGN requirements defined, ATIS has provided reference architecture diagrams which delineates the scope of work. The purpose of these reference architecture diagrams is to provide examples of the various interfaces that need to be defined and the accompanying functionality that would need to be supported.

Figure 3.3 below shows the basic NGN network reference architecture, which brings out some of the major interfaces that would need to be defined [47]. This includes

- Customer premise interface to the access provider network. For Wireline access, these interfaces could include DSL technologies, optical technologies (such as GPON or WDM-PON), and so forth.

- Interfaces between the network infrastructure provider and the service and application providers.
- Interface between service providers.

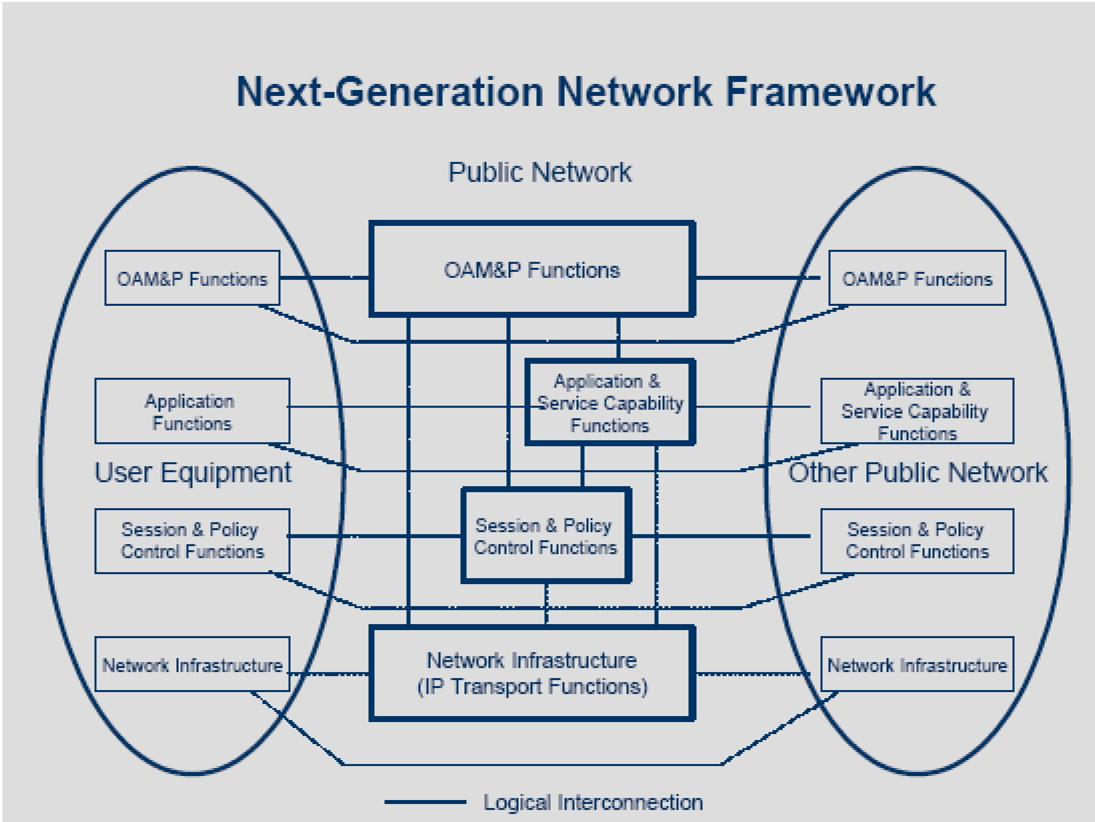

Figure 3.3 NGN Framework for a common architecture [35]

The functional components of the NGN framework shown above include the following

- **User Equipment** – it comprises of all the CPE including the private networks and terminal devices. At its simplest the user equipment could be a single device example, a mobile handset.
- **Other Public networks** – the NGN allows for the interconnection of multiple administrative domains, it is a means to provide interconnection and end to end communication even with non NGN networks such as traditional PSTN networks.

[35] ATIS, "Figure 4 - Framework for a Common Architecture," in *ATIS Next Generation Network (NGN) Framework, Part I: NGN Definitions, Requirements, and Architecture, Issue 1.0,* November 2004, pg 75

- **Public (NGN) Network** – the NGN network comprises four major functional blocks and they are defined below as given in [47].
    - **Network Infrastructure (IP Transport Function)** - The network infrastructure IP transport function provides for the transport of IP data, control, and management plane traffic to and from all elements of the NGN. This function also provides for the bearer-data, control, and management plane peer interworking of traffic to and from other networks where required. This includes public and/or private access, core networks, and interworking functions and devices.

    - **Session and Policy Control Functions** - This block provides the session and policy control functions that implement all aspects of session management including session establishment, session continuity, session modification, and session termination. Examples of functions contained within the control block are Session Control, Authentication, and Admission Control (administrative and resource constraint policy decisions) across all types of sessions – unicast, multicast, unidirectional, bi-directional, multi-connection.

    - **Applications and Service Capability Functions –** It supports application processes that can be invoked by the application subscriber or the application control to perform value added services for end users. Common services such as user database and presence are examples. This layer may also include media resources such as announcement systems, and media servers.

- **OAM&P Functions** – it contains Operations, Administration, Maintenance and Provisioning (OAM&P) functions required for an NGN. Fault, Configuration, Accounting, Performance, Security (FCAPS) functions are the basis for OAM&P.

The framework architecture shown in figure 3.3 is based on the NGN functional architecture being developed by ETSI TISPAN and the current view of the NGN functional architecture, its subsystems and the relationships between them; it is provided in their functional architecture Release 1[48]. Figure 3.4 below provides a high level view of the TISPAN NGN architecture, which is based on extended IMS.

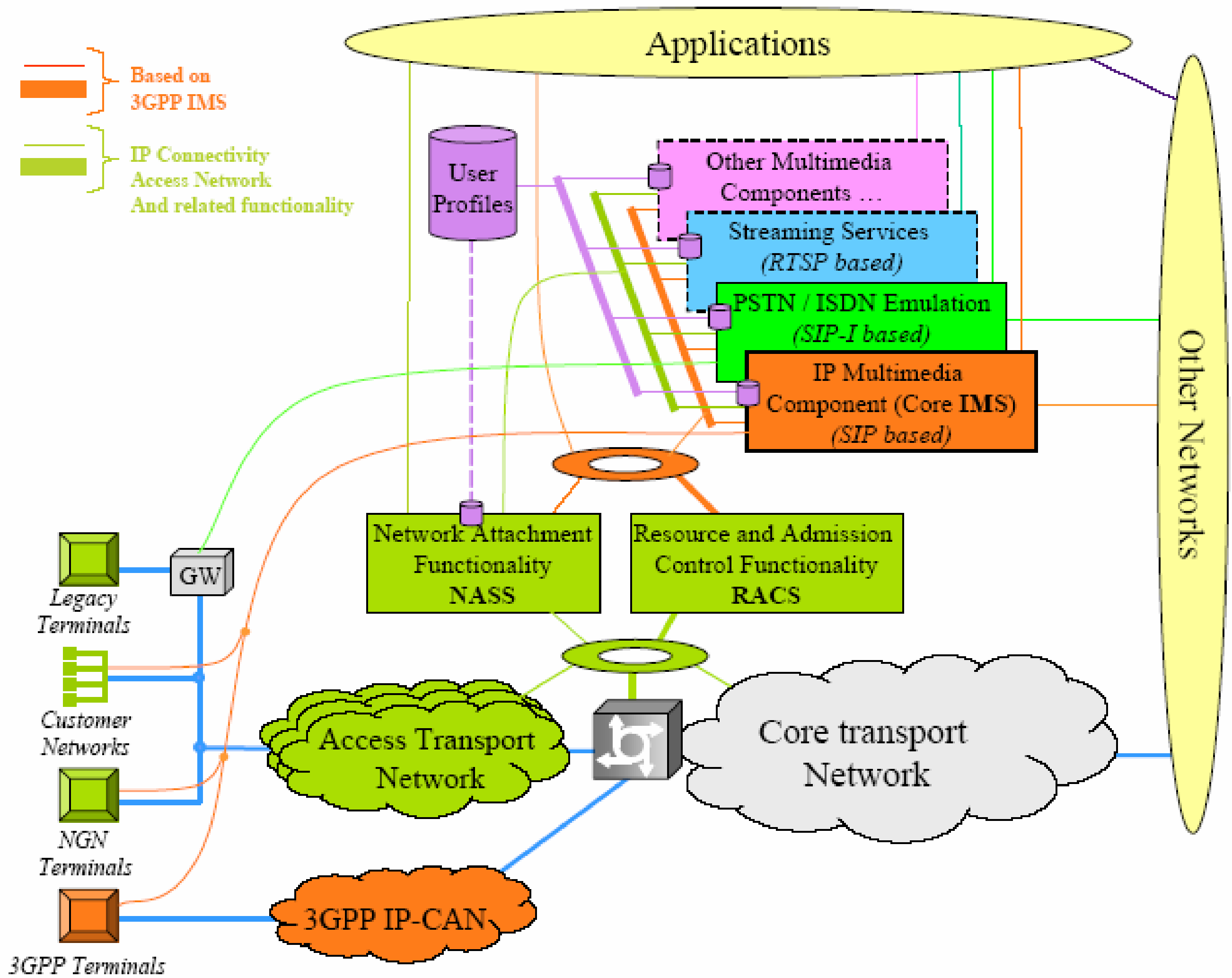

Figure 3.4 TISPAN view of the NGN Architecture[36]

TISPAN is working with 3GPP to extend the IMS to support additional access types. The existing IMS architecture is being modified to fulfill the NGN requirements. TISPAN is also working on defining the non-IMS components (e.g., NASS, RACS) as a part of TISPAN's NGN architecture. As this architecture is still being developed we will not examine it further, as very little information is available at present.

[36] ATIS, "Figure 5 - TISPAN View of the NGN Architecture," in *ATIS Next Generation Network (NGN) Framework, Part I: NGN Definitions, Requirements, and Architecture, Issue 1.0*, November 2004, pg 76

3.3.3 ATIS: Roadmap approach to NGN development

The approach adopted by ATIS to develop NGN systems includes identifying a roadmap. It is not a simple one due to the large number of activities that need to be performed by different parties. This is discussed in the beginning of part II of the NGN Framework [49], under the section which deals with assumptions and constraints around deployment of NGN.

The primary purpose of identifying the NGN Roadmap is the creation of an infrastructure that enables the flexible and efficient creation of NGN services. The focus of the roadmap is not on determining which services will be successful, as it's considered extremely risky to try to predict. Rather attention is being paid to identify the underlying network service enablers that will allow potential new services to be introduced. ATIS believes in letting the market decide which services will be successful rather than predicting which ones would.

This document also accepts the fact that the NGN is a process or an ongoing activity and so they are aware of the fact that target is moving and so the requirements and the Roadmap needs to be frequently revisited and revised. The Roadmap is being developed keeping in mind the heterogeneity in the industry in terms of technological (IP network capabilities) starting positions in moving towards NGN.

The network service enablers required for developing new services suggested as the starting point of the NGN include

- Unified user profile
- Security (various aspects as defined in NGN framework)
- Decoupling of services from access technologies
- Integrated management of all services, users and networks
- Presence
- Scalable management and operations
- Quality of Service (QoS)
- Settlement (accounting)
- Digital Rights Management (DRM)
- Media Resource Functions (MRF) etc

All these network service enablers have been assigned different priorities and have other functions and enablers dependent on them; details are provided in [49].

Figure 3.5 below illustrates the flow of new network service enablers into wider deployments. It needs to be noted that the NGN Service Capability Roadmap may not be the only relevant milestones, others may also be possible.

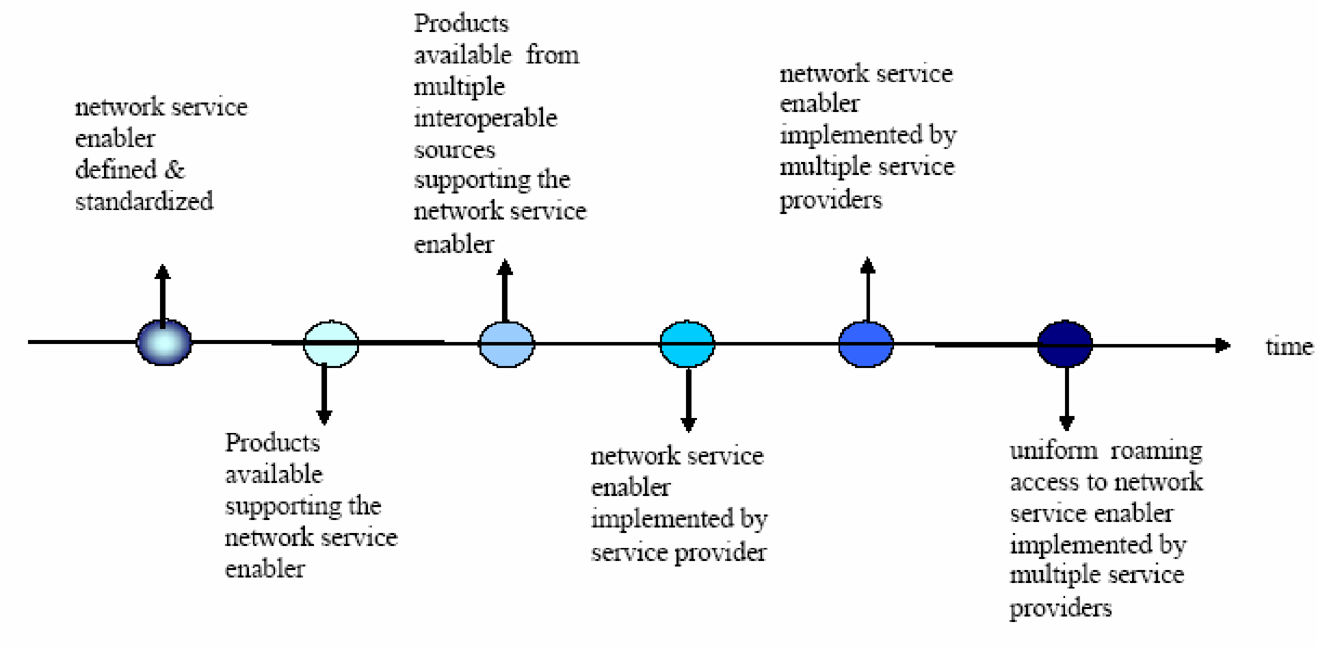

Figure 3.5 NGN Capability Deployment Roadmap[37]

As mentioned earlier the evolution of NGN is definitely not simple, it can evolve from service capabilities in various dimensions. Different service providers might want to adapt NGN along different paths, depending on their priorities, infrastructure, and business model, etc. Figure 3.6 below shows the various dimensions on possible NGN evolution [49].

Besides the traditional axes of NGN evolution there are other dynamic factors affecting the demands on the overall network infrastructure, such as market segment changes, emerging markets, universal access, enhanced client devices, new usage scenarios, different billing and charging methods, that do not fit into so simple a two axis model.

[37] ATIS, "Figure 2 NGN Service Capability Deployment Roadmap," in *ATIS Next Generation Network (NGN) Framework, Part II: NGN Roadmap 2005, Issue 1.0,* August 2005, pg 7

Traffic type vs. NGN function	Transport	Storage	Processing	Display	Content	**Other Axes**
Data	NGN					Security Privacy Charging Regulation Fads New Applications New Media Social Demands
Text						
Image						
Video						
Voice						

Figure 3.6 NGN Evolution in Different Dimensions[38]

So it can be seen that the development of NGN comprises of a large number of possible paths and the one adopted by ATIS takes into consideration the dynamically shifting industry requirements.

3.4 Need for Convergence and Next Generation Networks

So far we have seen the various desired capabilities of the future networks; these desired services and abilities are not new but have been envisioned in the past by people with foresight. There have also been attempts to create similar converged systems, which have been very complicated and economically non viable. But lately there have been certain advancements in technology as well as change in the market and customer requirements which have facilitated the development of such a converged system. In this section we

[38] ATIS, "Figure 3 Illustration of NGN Evolution in Different Dimensions," in *ATIS Next Generation Network (NGN) Framework, Part II: NGN Roadmap 2005, Issue 1.0,* August 2005, pg 8

will list a few of these changes in telecommunication industry which have provided a push towards rapid development of such advanced converged communication systems.

3.4.1 The next logical step

The development of any kind of technology usually follows a natural path of advancement. This path can be affected by a large variety of factors including the popularity of the services provided, the cost, the competition or the market forces. In the field of telecommunication, these forces are very active and have always forced the service providers to continuously develop new and improved systems and service which have been cheaper than their predecessors and of higher quality. These advancements are mostly evolutionary but sometimes they are revolutionary. The advancement in the telecommunication industry as described in the first two chapters has been steady but lately has picked up due to massive advancements in the supporting technology such as the microprocessor technology, improvements in digital media storage, better air interfaces to transmit information, greater infrastructure support for transport at the core etc. This has led to development and improvement in capacity of all forms of communication such as voice, data video, etc. and on all mediums of transport. But all these services grew in isolation, or vertically. So I believe the convergence of these systems to be a natural next step in the advancement of the way we communicate.

3.4.2 The internet: common platform for service deployment

The internet with its rapid growth has been considered one of the major reasons for advancement of the overall communication industry. As mentioned in chapter one, the internet usage has increased tremendously and has completely changed the way people communicate. It has been a complete paradigm shift. The common protocol (TCP/IP) has provided a common platform for services to be introduced with minimal effort and global reach. This goes hand in hand with the continuous increase in the speed of access and data transfer due to improved core network architecture as well as better end user delivery platforms such as cable, xDSL, WiFi, and WiMax (in the very near future).

The increase in the popularity, reliability and speed of the internet has allowed for new telecommunication services to be launched by nontraditional service providers. These communication services provide nearly the same quality with the additional benefit of flexibility, ease of set up and other such advantages which the internet provides. This has resulted in increased competition for the traditional service providers and has forced them to develop systems which allow them to compete with these new services. These systems use Internet Protocol, which is the only way they can compete.

So this shift in medium of voice delivery from pure circuit switched and analog to a combination of circuit and packet switched has forced a lot of development work in the technology in order to better utilize the existing infrastructure and at the same time allow for new technology to be introduced in the core. This has been a big push towards the development of NGN.

3.4.3 Need to contain the decreasing ARPU

Average Revenue per User (ARPU) is a very important instrument for measurement used by the industry to determine the extent to which their services have gained popularity or have become successful. Obviously the aim of each and every service provider is to increase the ARPU, by introducing new services and attracting more people to use nontraditional or data service packages being offered, especially by the wireless service providers. The increase in the ARPU which in turn results in increased profits allows the companies to pay for the costs of deploying advanced technologies and systems.

Intense competition has resulted in companies not being able to achieve the desired rates of increase in the ARPU, so other ways to achieve the results need to be introduced.

The converged system architecture allows for a higher level of integration and would result in an increase in the usage of the data service by the customers, thus increasing the ARPU. The industry is hopeful that the customers will be willing to pay more for services which make their lives easier and allow them to save in other areas. For example, advanced telecommunication will allow people to be able to stay in touch with their colleagues more effectively and have a higher flexibility in their work schedules; reduced commuting to work and working from home would be more convenient. Introducing a level of intelligence in the way in which communication services are provided, allows people to be more efficient and in turn would boost the revenues of the service providers. A large number of innovative services can be launched easily and quickly, which in turn would allow for more avenues to increase the revenue. So the need for increasing the ARPU has been a major reason for the push towards the development of NGN.

3.4.4 Changes in the structure of the telecom industry

The way in which telecommunication and entertainment services are provided has changed recently, especially in terms of the service providers and who provides which service. The ability to provide video, voice and data is available to both the traditional telecom companies such as SBC, AT&T, BellSouth, etc. and other communication providers including cable and wireless companies such as TimeWarner, Comcast, etc. Triple play services have essentially created a new market, and this has led to these different service providers being able to enter each other's market segments. Increase in competition has resulted in pressure on the companies to provide new and improved services at lower costs. The traditional technology or the vertical approach to application and service development requires a substantial amount of effort and money to introduce new services. One of the main objectives of the NGN is to allow for easy, quick and efficient introduction of new services; the common IP based converged architecture allows for this to be accomplished. So it can be stated that the change in the structure of the telecommunication industry and its transformation into a more general communications industry is a major driver for the development for NGN.

3.5 Conclusions

It is clear that the entire communications industry has realized the need for a change in the way services are provided. The industry leaders have come together to accurately identify the changes that need to be made; they have attempted to identify precisely the requirements of the Next Generation Networks. This is clearly visible by the efforts of ATIS and other similar organizations. They have also formulated a path to achieve those requirements and it is globally accepted to be based on the enhanced IMS architecture. Information about the development and standardization of the IMS architecture is provided in the following chapter.

4. IMS: General description

4.1 Introduction

The previous chapter dealt with the requirements of NGN and the steps being taken by various organizations to develop it. It is evident that the concept of convergence is the central idea behind those efforts. In this chapter we will examine the newly developed IMS architecture which forms the basis for the development of NGN systems.

IMS is being developed and standardized for years now and the standardization process is not simple by any means. It involves a number of agencies and organizations. We will start off by defining the roles of the different standardization bodies, their relationship and their joint effort to develop the various aspects of IMS. This will be followed by the description of the IMS architecture where the functionality of each network element will be defined and explained along with the protocols used in IMS.

4.2 IMS standardization process and organizations involved

IMS uses internet based protocols for performing various tasks and operations. This includes establishment of sessions, control, Authentication, Authorization and Accounting operations (AAA) etc. The task of the standardization bodies is to choose the appropriate protocol for a specific functionality, this may sound simple but its not. There are times when the existing protocols are not suitable or sufficiently powerful to support the IMS requirements. In such situations the standardization bodies need to work together to modify the existing protocols or develop an entirely new one.

A few of the standardization bodies were mentioned in the previous chapters, including 3GPP, 3GPP2, IETF (Internet Engineering Task Force) etc. It was also mentioned that ITU-IMT 2000 is the global standard for 3G networks and 3GPP and 3GPP2 are responsible for developing the required 3G standards. In this chapter we will pay attention to the activities of 3GPP and its relationship with IETF. We will start by looking at the structure and operation of IETF.

4.2.1 Internet Engineering Task Force (IETF): Structure and Operation

IETF is a loosely organized, self governed organization comprised of a wide variety of people with different technical backgrounds including network designers, vendors, etc. Their efforts are directed towards the development of the architecture, protocols, and the operations of the public internet. Their mission statement is documented in RFC 3935 [50].

The IETF is organized into a number of working groups and the actual technical developmental work is done under one of the working groups. The number of working groups is very large and they are not permanent; they are dissolved or rechartered once they have delivered their documents. These working groups are organized into Area Directorates; there are currently seven areas and all these are managed by the Internet Engineering Steering Group (IESG). The IESG is responsible for the technical management of the IETF and decides the area the IETF should work on. The members of the IESG also review all the specifications that are produced.

The process for developing standards for the internet is documented in RFC 2026 [51] and is briefly described here. As mentioned earlier, all the technical work is done in the working groups. The members of the technical groups comprise a number of volunteers and work independently as individuals. The interaction among the group members is via email and face to face discussions are held three times a year.

The technical documents used within the group are called Internet Drafts and they can be of two types, individual submission and working group items. The individual submission are reviewed by the members and becomes group item if found worthy of investigation by the rest of the group.

Work is done on the group item and is eventually submitted to the IESG, when the group is confident that it's ready for publication. The IESG provides feedback and approves the publication of a new RFC (Request for Comment). The internet drafts can be considered to be stable specifications only after they are RFCs.

There are three types of RFCs, Standards-Track, Non-Standards-Track and Best Current Practice (BCP) RFCs. All the RFCs published so far are available along with other useful information at the IETF website, http://www.ietf.org/home.html

4.2.2 3GPP: Structure and Operation

3GPP and 3GPP2 were introduced in chapter 2, as mentioned earlier; these organizations are responsible for the development of the 3G cellular protocols and standards. IMS was created as a means to merge the cellular and the internet world but it was further extended to include the traditional wireline communication systems including PSTN. In this section we will look at the structure of 3GPP and the process of standardization. 3GPP2 is similar in most aspects so we will not pay too much attention to 3GPP2.

We know that 3GPP comprises a number of international Standard Development Organizations (SDO) such as ARIB, ETSI etc. 3GPP was chartered to develop 3G standards and specifications based on GSM system. It led to the development of the UMTS-WCDMA system for 3G communication, which was an extension of the GSM, followed by GPRS system. This brings out a very important fact that the development of IMS is just one of the numerous activities undertaken by 3GPP and therefore the process of development of specifications is an exhaustive and elaborate task.

3GPP is organized into Technical Specification Groups (TSG) and they are managed by a supervising organization called the Project Co-ordination Group (PCG). Figure 4.1 below, as given at 3GPPs website (www.3gpp.com) shows the organizational structure [34]. The original six TSGs have been consolidated to four, and the work has been reassigned. The TSGs are further divided working groups as shown in the figure below and each of them is allotted particular tasks.

The TSGs do not produce standards but they deliver Technical Specifications (TS) and Technical Reports (TR). Once these are approved by the TSGs they are submitted to the organizational partners for the documents to go through their individual standardization processes.

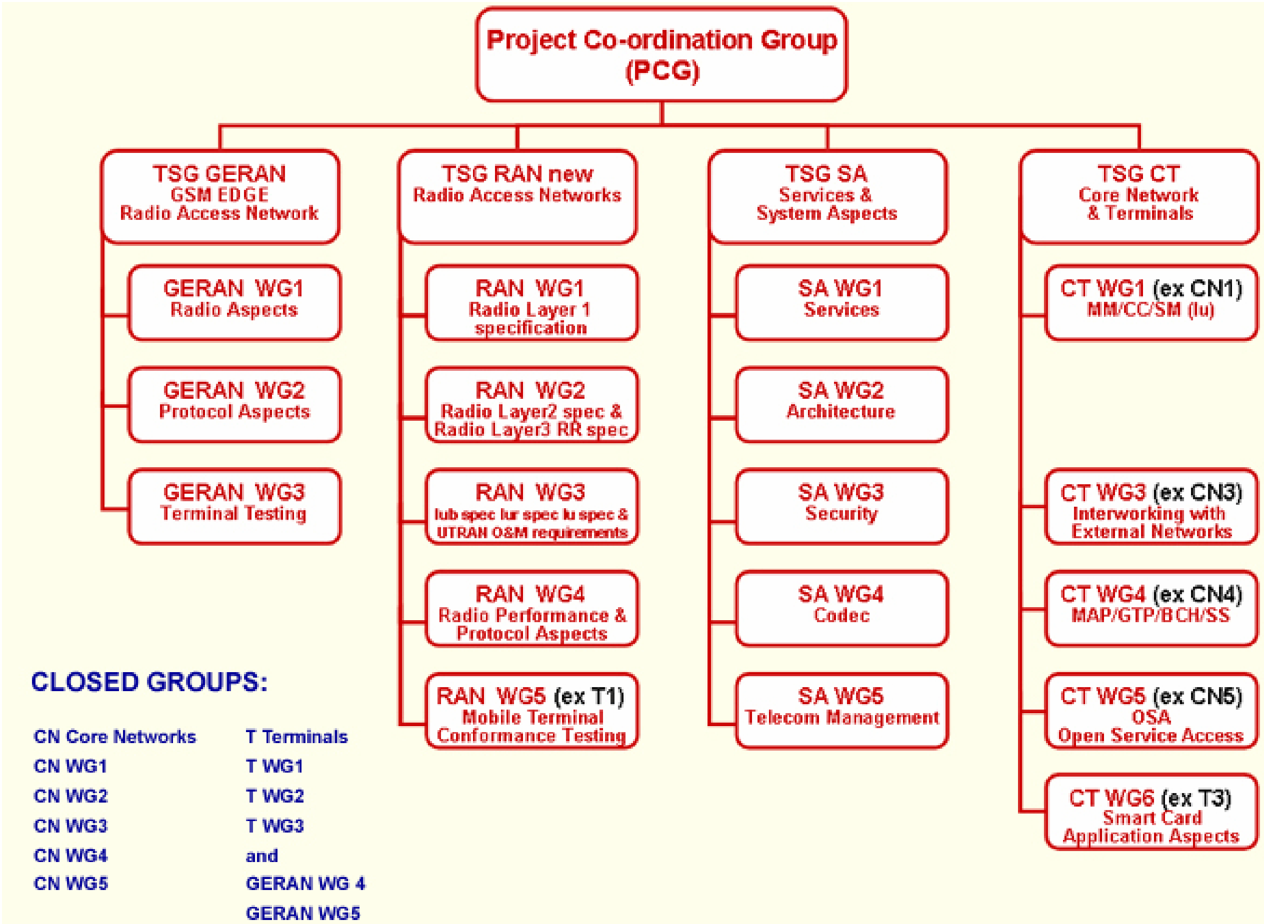

Figure 4.1 3GPP TSG Organization [39]

3GPP2 has a very similar structure and operates in pretty much the same way; 3GPP2 version of IMS is called Multimedia Domain (MMD). 3GPP2 is organized into 4 Technical Specification Groups (TSG). TSG-A deals with Access Network Interface, TSG-C focuses on CDMA2000® technology, TSG-S on Service and Systems aspects, and TSG-X on Intersystems Operations. 3GPP2 also delivers Technical Specification (TS) and Technical Reports (TR), but follow a different numbering pattern due to obvious reasons.

[39] 3GPP website, http://www.3gpp.org/tb/home.htm

4.2.3 IETF-3GPP/3GPP2 collaborative efforts

Almost all the protocols selected for IMS are originally from the internet domain. But many of these are neither completely suitable nor capable of fulfilling the requirements of IMS. There was an obvious need to modify the existing protocols to meet the requirements of IMS, which were very clear and well defined. The best way to go about it was working with IETF and jointly developing the required protocols, and so a collaboration was established between these organizations and documented in RFC 3313 [52] (IEFT-3GPP) and RFC 3131 [53] (IETF-3GPP2). Information about these collaborative efforts is given in the second chapter of [54], and is divided in three distinct areas: Internet, Operations and Management, and Transport area.

The focus of collaboration in the internet area was in the fields of IPv6 and DNS. One of the outcomes of these efforts was the development of RFC 3316 [55] by the IPv6 working group. This RFC provides guidelines for the implementation of IPv6 on cellular hosts; it allows the terminal to recognize a GPRS network and use IPv6, it behaves as a regular internet host while using a different wireless access like WLAN.

In the Operations and Management area the collaboration between 3GPP and the IETF was in the development and modification of the COPS[40] and DIAMETER[41] Protocols. A modified version of COPS was chosen to be used in the IMS called COPS-PR. The DIAMETER protocol is used as the base protocol and application are developed on it for a specific purpose. These applications and command codes are provided by the IETF in RCF 3589 [56].

---

[40] Common Open Policy Service (COPS) is used to transmit the policy related information between the Policy Decision Point (PDP) and Policy Enforcement Point (PEP)

[41] DIAMETER is an improvement on the previous RADUIS protocol used to transmit Authentication and Authorization on the internet.

In the transport area most of the work done is for the development of the Session Initiation Protocol (SIP). RFC 3261 [57] was a result of a lot of hard work, but this was not sufficient as the requirements of 3GPP were not being met and thus many extensions of the RFC were needed. To accomplish this IETF created a new working group called SIPPING, which collected and prioritized the SIP requirements and forwarded them to the SIP working group, where the actual work was done. This new process of collaborative effort to develop SIP is documented in RFC 3427[58].

4.3 IMS Architecture: General Description

In this section the basic architecture and working of IMS is presented. The description will be detailed enough to provide a good understand of the structure and the components of the system which work together to provide the functionality required in the NGN environment.

The first subsection will define the basic requirements expected from the IMS environment, in terms of services, user abilities and internetworking with other networks. This will be followed by the basic core network architecture and description of each node or element involved. A brief description of the user profile and identity will be provided at the conclusion of this section.

4.3.1 IP Multimedia Subsystem (IMS): Requirements

Chapter 3 outlines the requirements of the future or the Next Generation Networks which expects them to be all IP packet based system. It is clear that there is shift from the traditional Circuit Switched (CS) to Packet Switched (PS) environment for the delivery of multimedia services. IMS was developed for this very purpose by the cellular industry. The objective was to bring the internet and the cellular worlds closer and to utilize the existing packet based services being provided on the internet, which would increase profits.

The IMS comprises a Core Network (CN) which is a collection of signaling and bearer related network elements. These CN elements operate collectively to provide multimedia services to the end user. The IP multimedia services are based on the IEFT defined standards for session control and bearer control. The IMS terminal connects to CN via an IP-Connectivity Access Network (IP-CAN), which functions merely as a means to transport IP data. This allows IMS to achieve **Access independence,** as defined in 3GPP TS 22.228 V7.2.0 [59]

> The ability for the subscribers to access their IP Multimedia services over any access network capable of providing IP-connectivity, e.g. via:
>
> - 3GPP (UTRAN, GERAN)
> - Non 3GPP accesses with specified interworking (e.g. WLAN with 3GPP interworking)
> - Other non 3GPP accesses that are not within the current scope of 3GPP (e.g. xDSL, PSTN, satellite, WLAN without 3GPP interworking)

A few other important definitions provided in [59] are given below.

IM CN subsystem: (IP Multimedia CN subsystem) comprises all CN elements for the provision of IP multimedia applications over IP multimedia sessions.

IP multimedia application: an application that handles one or more media simultaneously such as speech, audio, video and data (e.g. chat text, shared whiteboard) in a synchronized way from the user's point of view. A multimedia application may involve multiple parties, multiple connections, and the addition or deletion of resources within a single IP multimedia session. A user may invoke concurrent IP multimedia applications in an IP multimedia session.

IP multimedia service: an IP multimedia service is the user experience provided by one or more IP multimedia applications.

IP multimedia session: an IP multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. IP multimedia sessions are supported by the IP multimedia CN Subsystem and are enabled by IP connectivity bearers (e.g. GPRS as a bearer). A user may invoke concurrent IP multimedia sessions.

The high level requirements of IMS are mentioned below; it's a list of the features that the architectural framework created for the delivery of IP multimedia services need to support. These requirements have been clearly listed and elaborated in chapter three of [54].

4.3.1.1 Support to establish IP Multimedia Sessions

IMS can provide the users with a variety of services but the most basic and most important service is audio and video communication. The IMS architecture allows the users to invoke multimedia sessions in a controlled and a flexible manner. It allows for usage of a large variety of media types and ensures interpretability. The IP multimedia session is designed to support one or more multimedia applications. The IMS systems also ensures that there no compromise or reduction in privacy, security, or authentication as compared with traditional systems.

4.3.1.2 Support for QoS (Quality of Service) negotiation and assurance

One of the main problems with internet based multimedia services is the lack of QoS. The IMS provides support for QoS negotiation for IP multimedia sessions, both at the time of establishment and during the session by the user and the operator. Same applies for individual media components. It also ensures end to end QoS for voice at least as good as that achieved by the circuit switched wireless systems (AMR based).

4.3.1.3 Support of interworking with the internet and CS domain

Support of interworking with the internet domain is most important and in a way obvious. The IMS users will be able to access information, services and applications available through the internet. This connectivity will provide a tremendous increase in the number of potential sources and destinations for multimedia sessions. The IMS users will have the ability to establish IP multimedia session with non IMS users from the internet and the existing circuit switched systems including PSTN and cellular networks.

4.3.1.4 Support for roaming

Roaming has been a requirement for cellular systems for some time now; IMS will allow users to roam between different service providers' networks. There are established procedures to transfer signaling, authentication, authorization, accounting, and other service related information between different IMS operators in a standardized and secure fashion.

4.3.1.5 Support for Service delivery control by the operator

The system requires strict control in terms of service delivery options. The operator will have control over all the services being offered to the users; these policies for control of user services have been classified into two categories.

General policies would be implemented throughout the network and would apply to all users; these restrictions might include control over the type of coded permitted for audio or video information. The purpose of such restrictions might be to monitor and control the bandwidth requirements in the network.

Individual policies on the other hand would apply to a particular user; they are configured specifically for each user depending on the subscription. The operator will have the ability to prevent a IP multimedia session from launching if the user is not authorized to use a type of media, for example video services.

4.3.1.6 Support for non standardized rapid service creation

The design of the IMS architecture has been influenced very strongly by this requirement. According to this requirement the IMS services do not need to be standardized. IMS would provide the necessary support for developing such services, or it would provide the necessary service enablers.

This allows the service providers or application developers to economically and rapidly develop and deploy services which would work equally well in different IMS networks. This is a radical change from the previous model of operation in the cellular field where almost all the applications were proprietary and did not always work in visiting networks (while roaming).

IMS has been able to successfully reduce the time and effort for developing a new service or application, by standardizing the service capabilities instead of the services.

4.3.2 IM CN Subsystem Architecture: Nodes and their functions

Before describing the IMS architecture, it is important to remember that IMS does not standardize a network element, but the functionality provided by the element. The manufacturer is free to decide about the physical design of the functional unit; two or more may be combined if deemed necessary. On the same lines, IMS does not standardize services but the service enablers, as mentioned in the previous section.

The core network elements or nodes are shown in figure 4.2 below. These nodes communicate with each other using specific protocols; each of these interfaces is identified using a reference point label as shown in figure 4.2. A detailed list of the all the interfaces and their operation is available in 3GPP TS 23.002 [63].

Also another important characteristic of the IMS architecture is that it exclusively uses IPv6; it requires network elements such as NAT-PT (Network Address Translation – Protocol Translation) and IMS-ALG (IMS Application Level Gateway) to interoperate with the traditional internet (which mostly uses IPv4). Presented below is a brief description of the nodes and their role in the IMS environment.

4.3.2.1 IMS Nodes : Databases

IMS utilizes one or more databases. The Home Subscriber Server (HSS) is used to store all the user related information, which is required to establish and handle multimedia sessions. The user information may include items such as the user profile (including the services the user subscribes to), location information, security information, the allotted S-CSCF address, etc. all the information is stored in a standard format and decisions are made by the HSS about the user sessions based on these items or user information. There could be more than one HSS, if the number of users is too high or for redundancy. In this case a Subscriber Location Function (SLF) is used to locate the HSS where the user information is stored.

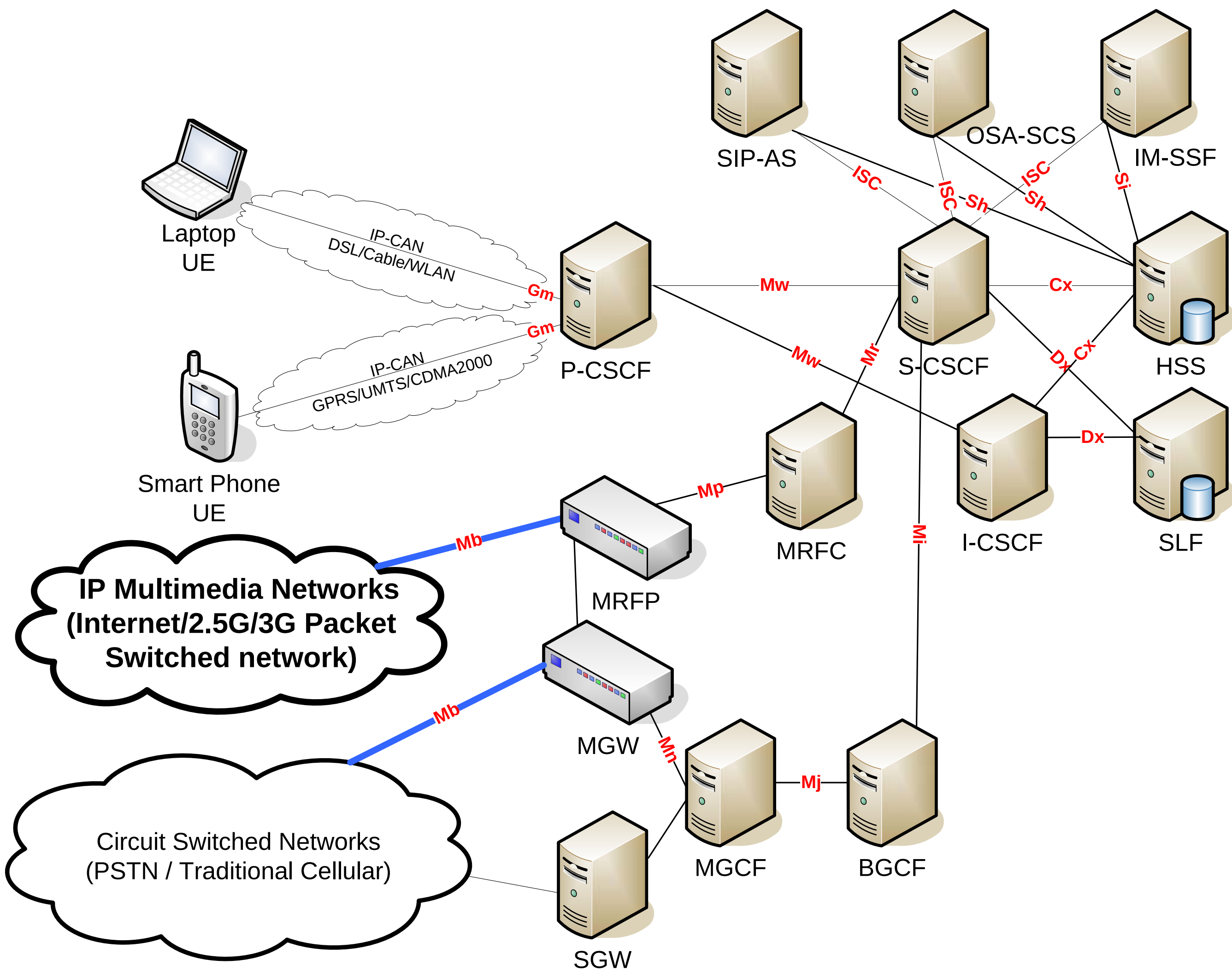

Figure 4.2 IMS Architecture General Overview

The SLF is a very simple data base which maps the user's address with an HSS, where all the user information is stored.

Both the HSS and SLF use DIAMETER protocol defined in RFC 3588 [60]. The DIAMETER protocol is the base protocol, there are specific applications developed for IMS to make the necessary decisions in the matters of authentication, authorization and accounting for a particular user.

4.3.2.2 Serving Call Session Control Function (S-CSCF)

The S-CSCF is a SIP based server and is one of the three types of Call Session Control Functions (CSCF). The basic function of a CSCF is to process the SIP (Session Initiation Protocol) signaling in the IMS. SIP is defined in RFC 3261 [57] and provides all the functionality to establish and manage multimedia sessions over IP networks. The S-CSCF is considered as the central node in the signaling plane. It is basically a SIP server but performs session control as well. It also maintains a session state as required by the network operator.

Within a network there could be a number of S-CSCFs, with different functionality and used for different purposes. The main functions of the S-CSCF as defined in 3GPP TS 23.228 [61] are mentioned below

Registrations related operations

- It may perform the task of the Registrar, as defined in RFC 3261 [57]. It accepts the registration requests from the users, verifies the request by downloading the authentication vectors from the HSS. DIAMETER protocol defined in RFC 3588 [60] is used for this purpose over the Cx interface.
- It downloads the user profile from the HSS, which contains the service profile or information about any application servers which need to be included in the SIP procedures.
- The S-CSCF makes the information available to the location servers, thus linking a particular user to a S-CSCF for the duration of the registration.

Session related and session-unrelated flows

- It controls the sessions for the registered users and might deny establishment of different sessions (IMS communication) on the basis of various conditions or clauses that bar such an activity for that particular user.
- The S-CSCF may behave as a proxy server as defined in RFC 3261[57] or subsequent versions of the protocol. It may accept and service requests locally or forward them to the relevant node after translation and filtering the request.
- The S-CSCF has the ability to behave as a User Agent as defined in RFC 3261 [57]; it can terminate and generate SIP transactions independently.
- It can interact with different service platforms or application server over the ISC (IP Multimedia Subsystem Service Control) interface. This interface allows for the coordination and support of various services provided by the application servers.
- The S-CSCF provides the endpoints with different service related information such as notification, location of additional media resources, billing notification etc.
- The functionality can be classified into two types; services provided for the originating end point and the for the destination end point. First we look at the services provided for the originating end point which may include an originating user/User Equipment or an Application Server (AS).
    - The S-CSCF obtains the address of the Interrogating CSCF (I-CSCF) of the destination user from the destination name. This is done in the case of

the destination user being the customer of a different network. The S-CSCF forwards the request to the corresponding I-CSCF.

  - If the destination user belongs to the same network the S-CSCF forwards the SIP request or response to the allocated I-CSCF with in the network.
  - The S-CSCF also forwards the SIP request/response to a SIP server that is not a part of the IMS (e.g., internet). This depends on the policies of the home network operator.
  - The S-CSCF forwards the SIP request or response to a Breakout Gateway Control Function (BGCF) for routing calls or sessions to the PSTN of any other Circuit Switched Domain (e.g. traditional cellular providers).
  - If the incoming request is from an Application Server (AS), the S-CSCF will verify the request coming from the AS is an originating request and proceed accordingly. The S-CSCF will process and proceed even if the user on whose behalf the AS is acting is not registered and reflect in the charging information that AS initiated the session on behalf of the user.

- Services for the destination endpoint (terminating user/UE)
  - The S-CSCF will forward the SIP request or response to a specific Proxy CSCF (P-CSCF) as a part of the SIP terminating procedure to a home user within a home network or for a roaming user in a visited network.
  - The S-CSCF will forward a SIP request or response to an I-CSCF as part of the SIP terminating procedure for a roaming user in a visited network, where the home network operator chooses to include the I-CSCF in the path.

- The S-CSCF will modify the SIP request as per directions for the HSS and the service control interactions, for routing the incoming session to the CS domain. This allows the user to receive the incoming session via the CS domain. It also forwards the SIP request or response to a BGCF for call routing to the PSTN or a CS domain.
- The SIP request might contain preferences for the characteristics of the destination endpoints, the S-CSCF performs preference and capability matching as specified in RFC 3312 [62].

Charging and resource utilization management monitoring

- Like all the other nodes of IMS, the S-CSCF is a part of the complicated charging or accounting procedure. It generates Charging Data Records (CDR) for this purpose.

The S-CSCF is always located in the home network, and there are usually a number of S-CSCFs in a network for the sake of scalability and redundancy. Each of them can serve a number of IMS terminals at the same time.

4.3.2.3 Proxy-Call Session Control Function (P-CSCF)

As shown in figure 4.2 the P-CSCF is the first contact between the user and the IMS network in the signaling plane. All the signaling and control information passes through the P-CSCF before getting to the user. It acts as an outbound/inbound SIP proxy server. The discovery of the address of the P-CSCF and its allotment to the user is performed during the process of IMS Registration and this does not change for the duration of the registration. The P-CSCF discovery process is described in section 5.1.1 of 3GPP TS 23.288 [61].

The various functions performed by the P-CSCF are mentioned below

- The P-CSCF establishes a security association between itself and the IMS terminal. The security association requirements and procedures are provided in 3GPP TS 33.203[77]. The IPsec security associations provide integrity protection. The P-CSCF also authenticates the user and asserts the identity with rest of the nodes in the network, to avoid redundant authentication requirements.
- The P-CSCF forwards the SIP register request from the UE to the appropriate I-CSCF determined from the home domain name provided by the user. This allows for the successful IMS registration of the user/UE.
- The P-CSCF forwards the SIP request or responses to and from the UE to the allotted SIP server, which could be an S-CSCF. The address of the S-CSCF would have been obtained by the P-CSCF as a result of the registration process.
- The P-CSCF performs SIP message compression/decompression for the purpose of reducing the size of the messages and thus reducing the reducing the transmission time and quicker session establishment.
- The P-CSCF may also include a Policy Decision Function (PDF). It performs the task of authorizing the bearer resources and performing QoS management over the media plane. The PDF may of may not be included in the same physical unit. Details about this function of the P-CSCF provided in 3GPP TS 23.207[64].
- Like the S-CSCF the P-CSCF also generates the CDR and forwards the information to the charging collection node.

An IMS network may have multiple P-CSCF for scalability and redundancy. The P-CSCF may be located in the home network or in a visited network.

4.3.2.4 Interrogating-Call Session Control Function (I-CSCF)

The I-CSCF is SIP proxy server; it is placed at the edge of the administrative domain of an IMS network. It is a point of contact for a connection destined to the a user who belongs to that network, or a roaming user currently located within the service area of that network operator. The address of the I-CSCF is listed in the DNS (Domain Name System) database and is made available when a SIP server follows the protocol for locating a SIP server for the next hop, the protocol is provided in RFC 3263 [65]. The functions performed by the I-CSCF, mentioned in [59] are as given below.

- During the registration process the I-CSCF assigns a S-CSCF for a particular user. The I-CSCF communicates with the HSS and SLF just like the S-CSCF over the Cx and Dx reference points. It uses the DIAMETER protocol. The user information is received by the I-CSCF and depending on the requirements; it assigns a S-CSCF to the user if one is not already allocated.
- The I-CSCF routes a SIP request from another network to the S-CSCF after obtaining the address of the appropriate S-CSCF from the HSS.
- The I-CSCF may encrypt certain parts of the SIP message which may contain sensitive information about the home domain; this functionality is optional and is called THIG (Topology Hiding Inter-network Gateway).

- Like other CSCFs, the I-CSCF also generates CDRs to be transmitted to the charging collection node.

4.3.2.5 Application Servers

An application server provides value added services and can be located in the home network or any third party network. It is essentially a SIP sever which hosts and executes various services. There are different modes of operation for an Application Server. It can operate in SIP proxy mode, SIP User Agent mode (terminating or originating), SIP Back-to-Back User Agent (B2BUA) mode etc.

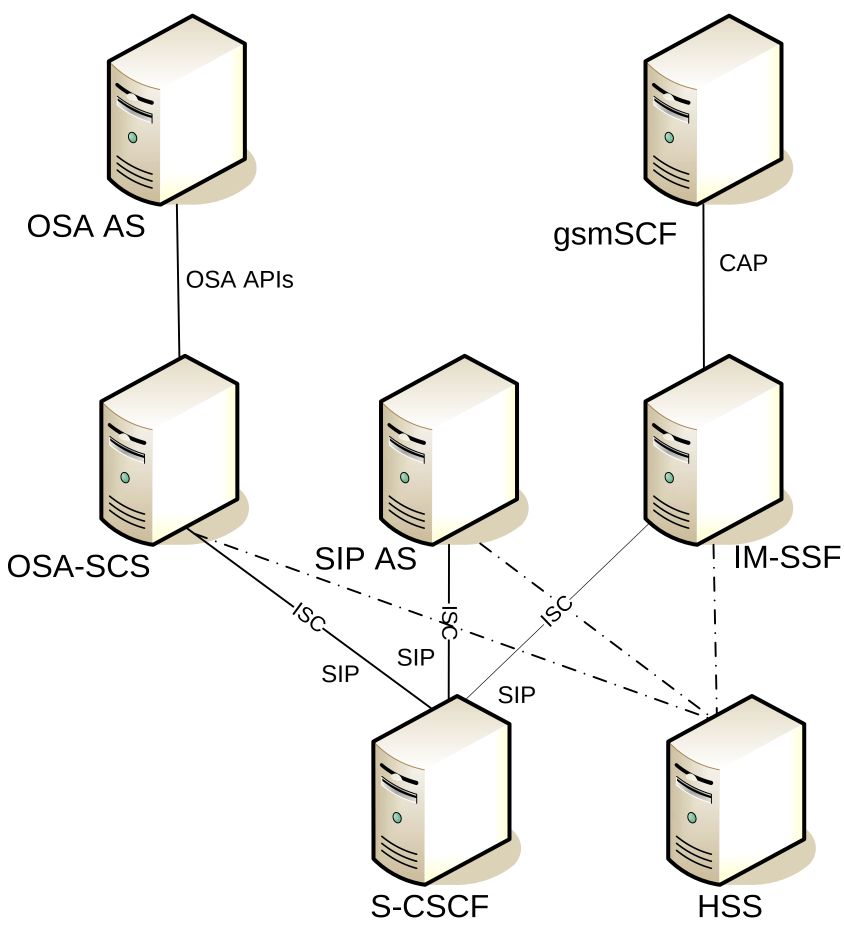

Figure 4.3 Types of Application Servers in IMS

The S-CSCF interfaces with the AS through the ISC (IP Multimedia Service Control) interface. The ISC interface is based on SIP [57]. Figure 4.3 shows three different types of Application Servers; they are described below.

- **SIP AS**: The SIP Application server is the native AS. It hosts and executes IP multimedia services based on SIP. All the new services that are going to be developed for the IMS architecture would be implemented using the SIP AS.
- **OSA-SCS**: The Open Source Access-Service Capability Server (OSA-SCS) provides an interface to the OSA framework applications. An AS located at a third party location will not be able to securely connect with the IMS network, whereas OSA has the capability to establish a secure connection with the IMS network. The OSA-SCS inherits all the abilities of OSA and is used to provide secure connectivity for a remotely located AS to the IMS network. The OSA-SCS acts as a regular AS and interfaces with S-CSCF via SIP on one end and as an OSA AS using OSA Application Programming Interface (API) on the other end. The OSA API is described in 3GPP TS 29.198 [66]
- **IM-SSF:** The IP Multimedia Service Switching Function is a specialized application server which allows for integration and reuse of the traditional applications developed for the GSM architecture. CAMEL (Customized Applications for Mobile network Enhanced Logic) was the name of the services that were developed to provide multimedia or enhanced services for GSM handsets. The IM-SSF acts as an application server on one side interfacing with the S-CSCF using SIP, and on the other side it acts as a Service Switching Function (SCF) interfacing with the gsmSCF with a protocol based on CAP

(CAMEL Application Part). The CAP protocol is defined in 3GPP TS 29.278 [67].

The three AS mentioned above may perform different tasks but they behave exactly the same towards the IMS network. They all appear as SIP AS behaving in one of the earlier mentioned modes.

The Application Servers present in the home network may optionally interface with the HSS. The SIP AS and the OSA-SCS interface with the HSS using DIAMETER protocol [60] and the interface is labeled ‘Sh’. The IM-SSF interfaces with the HSS using protocol based on MAP (Mobile Application Part) defined in 3GPP TS 29.002 [68].

4.3.2.6 Breakout Gateway Control Function (BGCF)

As shown in figure 4.2, the BGCF provides connectivity to the Circuit Switched domain through the MGCF (Media Gateway Control Function), SGW (Signaling Gateway) and the MGW (Media Gateway). These three nodes put together are referred to as the PSTN/CS Gateway.

The BGCF is basically a SIP server which has the additional capability of routing and establishing sessions based on telephone numbers as user addresses. The BGCF is used exclusively for sessions initiated by an IMS user who needs to communicate with a user in the PSTN or PLMN (Public Land Mobile Network) domain, both of which are in the Circuit Switched domain.

The main functions performed by the BGCF are as follows:

- It receives a request from the S-CSCF to select the appropriate PSTN/CS Domain break out point for a particular session.

- The BGCF selects the network in which the internetworking with the PSTN/CS Domain is to occur. If the interworking with PSTN/CS domain is to occur in the same domain, it selects the appropriate MGCF and forwards the SIP signaling to it.
- If the interworking with the PSTN/CS domain is to be done at a different network, the BGCF forwards the SIP information to the BGCF of that network. If network hiding is required, the BGCF will forward the SIP signaling through the I-CSCF to the other BGCF.
- The BGCF also generates CDRs to forward to the charging collecting node.

4.3.2.7 Public Switched Telephone Network / Circuit Switched (PSTN/CS) Gateway

The PSTN/CS gateway comprises of three components as mentioned earlier. The functions performed by each of them are given below.

MGCF (Media Gateway Control Function) interfaces with the BGCF and receives the SIP signaling. Its function is to convert the SIP signaling to either ISUP (Signaling System 7) defined in ITU-T Recommendation Q.761 [69] over IP or BICC (Bearer Independent Call Control) defined in ITU-T Recommendation Q.1901 [70] over IP. The converted signaling is forwarded to the Signaling Gateway (SGW). The MGCF also controls the resources in the Media Gateway (MGW). The MGCF and the MGW communicate with the help of the H.248 protocol, specified in the ITU-T Recommendation H.248 [71] (this protocol has been revised a number of time and annexes have been added to the main body).

SGW (Signaling Gateway) provides the signaling interface with the circuit switched domain. Its main function is to perform lower level protocol conversion. It converts MTP (Message Transfer Part) defined in ITU-T Recommendation Q.701 [72] into SCTP (Stream Control Transmission Protocol) defined in RFC 2960 [73] over IP. So the signaling format ISUP or BICC over MTP is transformed into ISUP or BICC over SCTP/IP.

MGW (Media Gateway) connects the media plane of the PSTN or any other CS environment with the media plane of IMS. The MGW transcodes the IMS data transported over RTP (Real Time Protocol) defined in RFC 3550 [74] into PCM (Pulse Code Modulation) used in the PSTN environment. Also the MGW performs transcoding in situations where the IMS terminal does not support the codec being used by the CS side.

4.3.2.8 Media Resource Function (MRF)

The Media Resource Function (MRF) handles all the media transportation and processing requirements. It is divided into two functional components as shown in figure 4.2, the Media Resource Function Controller (MRFC) and the Media Resource Function Processor (MRFC).

The **MRFC** interfaces with the S-CSCF over the Mr interface and uses SIP [57] for signaling purposes. The tasks performed by the MRFC, mentioned in [61] are as below

- It controls the media stream resources in the MRFP.

- The MRFC interprets the information forwarded by the S-CSCF and the Application Servers and modifies the operation of the MRFP according to the directions.
- The MRFC generates CDRs like the other nodes in IMS to be forwarded to the charging collecting node.

The **MRFP** is controlled by the MRFC though the Mp interface, also called a reference point. The Mp reference point does not have a specific protocol specified for it yet and has an open architecture to allow extension work to be carried out. It completely supports the H.248 Standard [71]. The tasks performed by the MRFP are given below as defined in [61]:

- The MRFP controls the bearer plane on the Mb reference point.
- It provides the functionality of mixing various incoming media streams in case of a conference call.
- It acts as a source of media streams or plays streams as for multimedia announcements.
- The MRFC performs all other media processing functions such as transcoding, media analysis etc.
- It also provides floor control or manages access rights in a conference environment.

4.3.3 IMS: Protocols Used

We know that the protocols used in the IMS environment are derived from the internet and wireless domain (GSM/GPRS). 3GPP decided to use the protocols being developed by the IETF and ITU-T for the IMS environment and thus was able to capitalize on their expertise in designing robust protocols and at the same time reducing the standardization time and costs involved.

Mentioned below is a brief description of the various protocols selected or developed to be used in different area of IMS.

4.3.3.1 Session Control in IMS

A lot of information has already been provided about the protocol of choice for the purpose of session control in the IMS environment. The protocol used for session initiation and control in IMS over IP networks is the Session Initiation Protocol (SIP) specified by the IETF in RFC 3261 [57]. SIP is a text based protocol unlike other session protocols such as BICC and H.323. This makes it easier to debug, extend, and build services on it. One of big reasons for choosing SIP was the fact that it is based on many familiar and successful protocols such as SMTP (Simple Mail Transfer Protocol) and HTTP (Hypertext Transfer Protocol). Also SIP follows the familiar client-server model.

SIP makes is very easy to develop new applications, which is one of the requirements of IMS.

4.3.3.2 Authentication Authorization and Accounting (AAA) in IMS

Authentication Authorization and Accounting (AAA) operations play a very important role in any network, especially in the IMS environment. It is of great importance to have an efficient and highly reliable mechanism to perform the tasks of authenticating a user's identity, authorizing the user to access the appropriate resources and making sure the resources and services consumed are logged accurately and billed correctly. IMS uses the DIAMETER protocol to perform the AAA operations. It allows different nodes to access retrieve or modify user information from HSS or SLF. The DIAMETER protocol is an improvement over the older RADIUS protocol (RFC 2865 [75]). The DIAMETER protocol is defined in RFC 3588[60]. The DIAMETER protocol is used over different interfaces such as Cx, Dx and Sh. It consists of a base protocol and is used to develop various DIAMETER applications. These applications are extended and customized for a particular purpose or an environment. The different interfaces may use different DIAMETER applications to perform the various AAA procedures.

4.3.3.3 Quality of Service in IMS

The QoS requirements for IMS were mentioned earlier in section 4.3.1.3, where we see that IMS requires the ability to negotiate and assure a certain Quality of Service in order to provide reliable IP based communication.

Generally there are two models to provide QoS on the packet switched IP domain specifically the internet. They are the Integrated Service model and the Differentiated Service (DiffServ) model.

Integrated Service model is defined in RFC 1633 [83], it provides end to end QoS. The endpoints request a certain QoS and the network grants it. The protocol used by the Integrated Services architecture is RSVP (Resource reSerVation Protocol), it is specified in RFC 2205 [84] and has been updated by RFC 2750 [85] and RFC 3936 [86].

Integrated Service works well in small networks does not scale well as the routers have to store state information about every flow and perform lookup before routing any packet.

The second model of QoS solves some of the problems faced while using the Integrated Service model. The DiffServ architecture is specified in RFC 2475 [87] and RFC 3260 [88]. The DiffServ servers need to maintain minimum state information about the flows and enables a quicker treatment for the packets flowing through them. In this architecture the router is aware of the treatment that needs to be given to each packet; the treatment is referred to as the Per Hop Behavior (PHB). Each PHB is identified by 8-bit codes called Differentiated Service CodePoints (DSCP). The DSCP information is carried by the packets in their IP headers. In IPv4 it is placed in the 'Type of Service' field and in IPv6 it is placed in the 'Traffic Class' field.

IMS allows many different end-to-end QoS models. All the models are described in 3GPP TS 23.207 [64]. The terminals may use link layer resource reservations methods such as PDP context reservation, or directly use protocols such as DiffServ or RSVP.

The IMS networks use DiffServ and may use RSVP.

4.3.3.4 Security in IMS

Security generally deals with integrity, confidentiality, and availability. There are various means to achieve the security requirements in the SIP environment. In IMS security can be divided in two different areas, Access security and Network security.

Assess security deals with authentication and authorization processes and establishment of the IPsec security authorization (architecture defined in RFC 2401 [76]); these procedures are performed during the REGISTER[42] transaction. All the procedures for security access are provided in 3GPP TS 33.203 [77]. In the 3GPP networks the user

[42] REGISTER transaction is a part of the SIP process

identity is stored on a smart card inserted in the IMS terminal; this card is usually known as UICC (Universal Integrated Circuit Card).

Network security deals with protecting the traffic between two nodes. The nodes may or may not belong to the network. There may be different levels of requirements from the network security mechanisms in place. If two different security domains are involved, the traffic travels through two Security Gateways (SEG). In this case the traffic is protected using IPsec ESP (Encapsulated Security Payload), specified in RFC 2406 [78] and runs in tunnel mode. The security associations are established and maintained using IKE (Internet Key Exchange), specified in RFC 2409 [79]. All the network security requirements are mentioned in 3GPP TS 33.210 [80]. Recently newer versions of the internet security protocols have been introduced (December 2005) but they are not being used in the latest IMS release (Release 7).

4.3.3.5 Policy Control in IMS

Policy control deals with the media-level access control; the decisions made by the policy control mechanism authorizes a user to use the media plane and assigns the QoS to be provided for that user session.

The media-level policy is enforced by the routers present in the network, but these routers do not have the ability to make decisions about users as they do not have access to the user information stored in the HSS. The task of obtaining the user information and making these decisions is performed by a SIP server in this case. The SIP server informs the routers to allow of deny a certain user with the requested media resources.

The node which makes the decision, in this case the SIP Server is called the Policy Decision Point (PDP) and the router is called the Policy Enforcement Point (PEP).

The protocol used between the PDP and PEP is called Common Open Policy Service (COPS) protocol, it is defined in RFC 2748 [81] and has been updated by RFC 4261 [82], which provides a higher level of security at the transport level.

COPS generally supports two models for policy control, the outsourcing model and the configuration model (also called provisioning).

In the outsourcing model the PEP contacts the PDP for every decision, whereas in the configuration model the PEP stores the policy from the PDP locally and uses it to make decisions. IMS uses a combination of the two models; it's called COPS-PR. It's a mixture of the two models as it uses the same message format and the Policy Information Bases (PIB) as used by the provisioning model and the policy decision are transferred in real time like in the outsourcing model.

In IMS there are two types of limitations on the session that can be established. They are user-specific limitations and general network related policies.

The user-specific limitations include restrictions on a particular user, in terms of resources that are allowed. An example would be an audio only subscription, so the user will not be allowed to establish video sessions.

The general network policies would apply to all the users of that network. This might include restrictions on the codecs that can be used.

The P-CSCF deals only with the enforcement of the general network policies, whereas the S-CSCF handles both user-specific policies and the network policies. Both these PDPs use the same mechanism to monitor the sessions. They access the SDP (Session description Protocol) body to identify the type of session and media requested during SIP procedures.

4.3.3.6 Media Transport in IMS

There are a large number of codecs available to encode audio, video, and data. Ideally there should have been a common set of codecs which would allow all kinds of terminals belonging to different networks to be able to communicate with each other without the need for any transcoders. But unfortunately 3GPP and 3GPP2 were not able to agree on a common codec and require transcoders to communicate between their IMS (MMD in case of 3GPP2) terminals.

3GPP and 3GPP2 individually do specify the codecs which will be supported by all their terminals. 3GPP terminals support AMR speech codec and the H.263 video codec; details are provided in 3GPP TS 26.235 [89]. It also supports AMR-WB (AMR-Wide Band) for terminals providing wide band services and for real time text ITU-T recommended codec T.140 [90] is used.

In IMS unreliable media transport is provided by using Real Time Protocol (RTP), defined in RFC 3550 [78] over UDP (User Datagram Protocol). UDP is the unreliable transport protocol defined in RFC 0768 [91]. RTP is used in conjunction with RTCP (RTP Control Protocol), which allows quality of service information and allows for inter media synchronization. This is essential in case of an unreliable media transport as packets might arrive in random order and a few might even be lost or dropped. RTP provides mechanisms to overcome such problems.

An alternative for providing unreliable transport is DCCP (Datagram Congestion Control Protocol). It's a new protocol being developed by the IEFT, and it may be used in the future but is not mature enough to be implemented on a large scale.

For reliable transport of media there are two protocols that are most popular; TCP (Transmission Control Protocol) and SCTP (Stream Control Transmission Protocol). In case of IMS, TCP would be natural choice for reliable media transport because of its proven track record and global acceptance.

4.3.4 Addressing and User Identification in IMS

Addressing and routing operations performed in the IMS environment are based on IPv6 architecture. The mechanisms which deal with these operations to provide assess to IMS services and general IP address management are provided in 3GPP TS 23.221 [92]. In this section we will briefly describe the methods and systems for identification of users in the IMS environment. The need to identify a user distinctively is universal requirement for any network; the same is true in IMS. There are various identities associated with every user in the IMS architecture. The following sections identify and define the purpose of these identities as defined in the 3GPP TS 23.228 [61].

4.3.4.1 Private User Identities

Each user is assigned a Private User Identity by the service provider. The purpose of the Private User Identity is for Authentication, Authorization, Accounting (AAA), and registration purposes and not for the routing or establishing sessions. The Private User Identities take the form of a Network Access Identifier (NAI) as defined RFC 4282 [93].

The format of a NAI is ‘username@operator.com’. The contents of the NAI for the private identity may include the IMSI (International Mobile Subscriber Identifier) in the case of GSM/GPRS/UMTS systems. IMSI is included on the UICC (Universal Integrated Circuit Card) used in 3G systems.

The purpose and utility of Private User Identities are mentioned below:

- Private User Identity is to be included in the all the registration requests passed from the User Equipment (UE) to the home network.
- The Private User Identity shall be securely stored on the ISIM (IMS SIM), and the user will not have the ability to modify it.
- The Private User Identity identifies the subscription and not the user; it is a globally unique identity defined by the home network operator.
- The Private User Identity is allocated to the user permanently for the duration of the subscription; it’s not a dynamic identity.
- The Private User Identity is used to identify the user information for a particular subscriber at the HSS; also it may be present in the charging records depending on the operator policies.
- The Private User Identity is authenticated only during registration of the user.
- The HSS stores the Private User Identity and is also required by the S-CSCF upon registration and deregistration.

4.3.4.2 Public User Identities

Public User Identities unlike Private User Identities are used to identify the user, for requesting communications with other users, etc. The Public User Identities are assigned by the operator and each user will be usually assigned two or more Public User Identites (details provided in 3GPP TS 22.228 [59]). The Public User Identities will be either a SIP URI (Uniform Resource Identifier), as defined in RFC 3261 [57] and RFC 3986 [94], or a TEL URI, defined in the RFC 3966 [95]. A few functions, uses and characteristics of the Public User Identities are given below:

- The Public User Identities are used as contact information of a user and can be distributed to others, on a business card for example.
- As mentioned earlier both numbering and internet names can be used for this purpose. The Public User Identity using a SIP URI will generally have the format of 'sip: first.last@operator.com' and the one using a TEL URL will have a format which includes a '+' sign followed by the complete number, an example is 'tel:+1-292-234-2343'.
- The ISIM (IMS SIM) can store one or more Public User Identities, securely and the user will not have the ability to modify it.
- The option for multiple Public User Identities is to facilitate the distinction between identities for different purposes, for example one might be for personal use, which is known by family and friends and another for work related contacts.
- IMS has a system which allows Public User Identities to be registered implicitly or explicitly. This procedure allows for a number of identities to be registered using just one Register request, thus saving time and bandwidth. All the Public identities can be registered independently as well.
- Public User Identities are not authenticated by the network during registration, and they can be used to identify the user information stored in the HSS.

4.3.4.3 Relationship between Public and Private User Identities and Service Profiles

The IMS operator assigns the various Public User Identities to the user. These Public User Identities are linked to one or more Private User Identities. All the information about the users along with other service related information is stored in the HSS in the form of Service Profiles. The structure and functionality of a user service profile as stored on the HSS is provided in 3GPP TS 29.228 [96]. The service profile includes a list of Public User Identities and a number of initial filter criteria. These initial filter conditions provide basic information about the user /operator preferences and usually lead to the invocation of one or more Application Servers.

Figure 4.4 below shows the relationship between the Public and Private User Identities along with the service profiles.

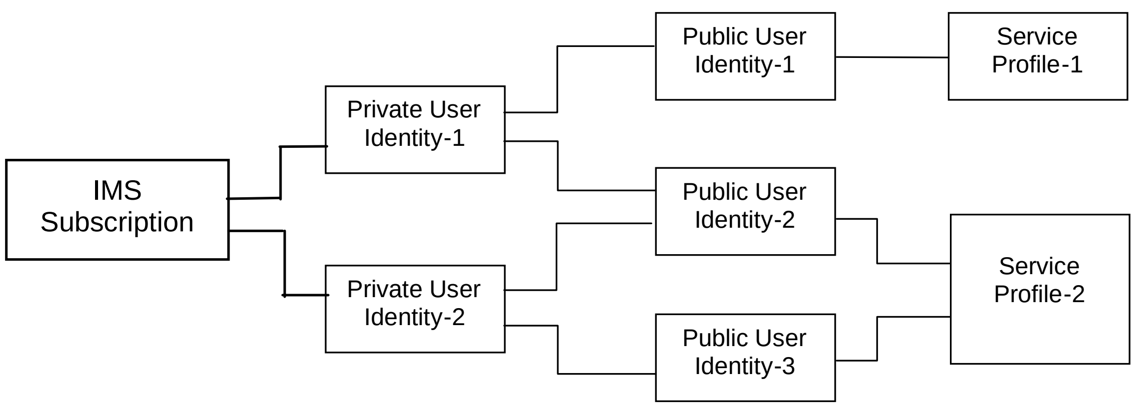

Figure 4.4 Relationship between Public and Private User Identities[43]

The scope of a Service Profile defined and maintained in the HSS is limited to the IMS environment. A service profile is downloaded by the S-CSCF from the HSS during registration, and only one service profile is associated with a Public User Identity at a given time.

[43] 3GPP TS 23.228 V7.1.0 (2005-09), page 27

As shown in the figure above a Public User Identity can be associated with just one Service Profile but a Service Profile may be associated with one or more Public User Identities.

A very important modification was introduced in release 6 of IMS. A Public User Identity was allowed to be shared among multiple Private User Identities. This allows a user to be able to access a Public identity while registered in two or more Private User Identities. An example of this situation would be a user accessing the same email account while being registered on two different devices. This is depicted in figure 4.4 above; Public User Identity-2 is associated with Both Private User Identity 1 and 2.

4.3.4.4 Public Service Identities

Public Service Identities (PSI) are used to define services that are hosted by different Application Servers (AS). The PSIs do not identify users, but they may identify groups. An example would be an application server hosting a chat group; it can be identified using a PSI. This PSI can be used to establish sessions and enable communication among the various subscribers of the service. A PSI can use the format of a SIP URI or a TEL URL, as defined section 13.5 of 3GPP TS 23.003 [97].

IMS allows for users to create, manage and use Public User Identities, running on an Application Server (AS). The creation of a PSI can be done both statically and dynamically.

The Public Service Identities do not have an associated Private User Identity, as in the case of a Public User Identity. This is so because a PSI does not require authentication.

4.4 Conclusions

This chapter provided a brief overview of the requirements and the capabilities of the IMS architecture and the various components which achieve these requirements. It was important to understand the process of IMS standardization to appreciate the scope of the system and its operation. As mentioned earlier the main purpose of developing this system is be able to provide advanced, value added, IP based multimedia services, which are converged in nature or span multiple systems. A few applications and services that have been developed include presence related services, instant messaging, push to talk services, etc. The services which can be developed on the IMS platform are limited only by the creativity of the application developers and requirements of the customer. IMS has provided a very large pool of service enablers which makes it easy and quick to develop new applications and build on existing ones. At the same time, being network agnostic it would allow for integration of advanced transport systems (both wired and wireless sectors) and other future developments in communications industry.

5 Convergence in the residential Environment using IMS

5.1 Introduction

The concept of convergence has been introduced and explained in the previous chapters. Here we will extend that concept and apply it to the residential environment. The concept of convergence in the residential environment has very close ties to the idea of residential automation, or the model of a digital smart home. There have been different views about the ways to achieve the requirements of a smart home and the requirements vary as well. In this chapter I will define the possible requirements of a networked (automated) home, specifically in the areas of communication systems and their inter operation. These requirements might not reflect the needs of an average person; it's a view for the near future.

After describing the requirements of this system, I will present a way of achieving it using the enhanced IMS architecture. This could be seen as a specific application of the IMS architecture.

5.2 Description of the converged residential environment

There has been a massive shift in the way communication services are provided. Bundling up of communication services has been very successful. Companies have been providing internet, voice telephony, and digital entertainment services to residential subscribers, with the convenience of a single bill. These services might include cellular telephony soon.

The idea behind the bundling up of communication services is to provide the customers with a convenient way to manage their communication services. What lacks in this environment is interworking among the various services. There have been some efforts to provide interworking but it has been very limited.

There are a number of value added services that could be introduced if there were a means to integrate all the multimedia services being consumed in the residential environment.

A converged environment would allow for rich multimedia applications to be accessed by the users on any device and retain the user profile and other settings. The idea is to be able to communicate, establish multimedia sessions, and perform control and configuration operations on all the networked elements in the residential environment.

The following are the expectations of the converged residential network:

- The system would allow multiple user profiles with different levels of control over services and the systems.
- This system would allow the integration of wireline and wireless (cellular) voice communication, or would allow the users to make and receive phone calls on wireline based digital/analog phone and the wireless cellular phone interchangeably as per personal preference.
- The system would allow for multimedia sessions to be established between the various terminals (audio/video database sever, TiVo etc) in the network. The users should be able to access the content with in the residential environment on any terminal or from outside the residential environment (restricted only by the capabilities of the device and the available bandwidth).

- The nodes in the residential environment may use either wired or wireless connectivity. The local area network should support secure wired and wireless access (WiFi or WiMax).
- The network must support application or service hosting and the application severs should be remotely manageable. There applications may include but are not limited to residential security system (monitoring, configuring and authorization), web hosting services, residential power/gas monitoring and control, etc.
- The residential environment should be secure, easy to manage, and efficient, and should provide effective means of managing and configuring devices in the network.

Figure 5.1 below shows such a converged architecture which shows a few of the many possible elements in the network. All these elements should be accessible to the user with sufficient rights, both locally and remotely. The users in the residential network subscribe to a number of services which might include cable television (digital or analog), Internet access, telephone connection, and one or more cellular phone subscriptions. As shown in the figure we will assume these services are being provided by the Multimedia/Communication service providers. All the elements in the residential environment are networked as in a LAN environment. All there services are controlled by a so called residential server which performs the control operation and a wireless router which provides the connectivity. There could be other systems operational in the residence, such as a security system, an electronic power supply and control mechanism, which controls the lights, air conditioning and other energy related functions, etc. All these are linked and controlled by the residential server as well.

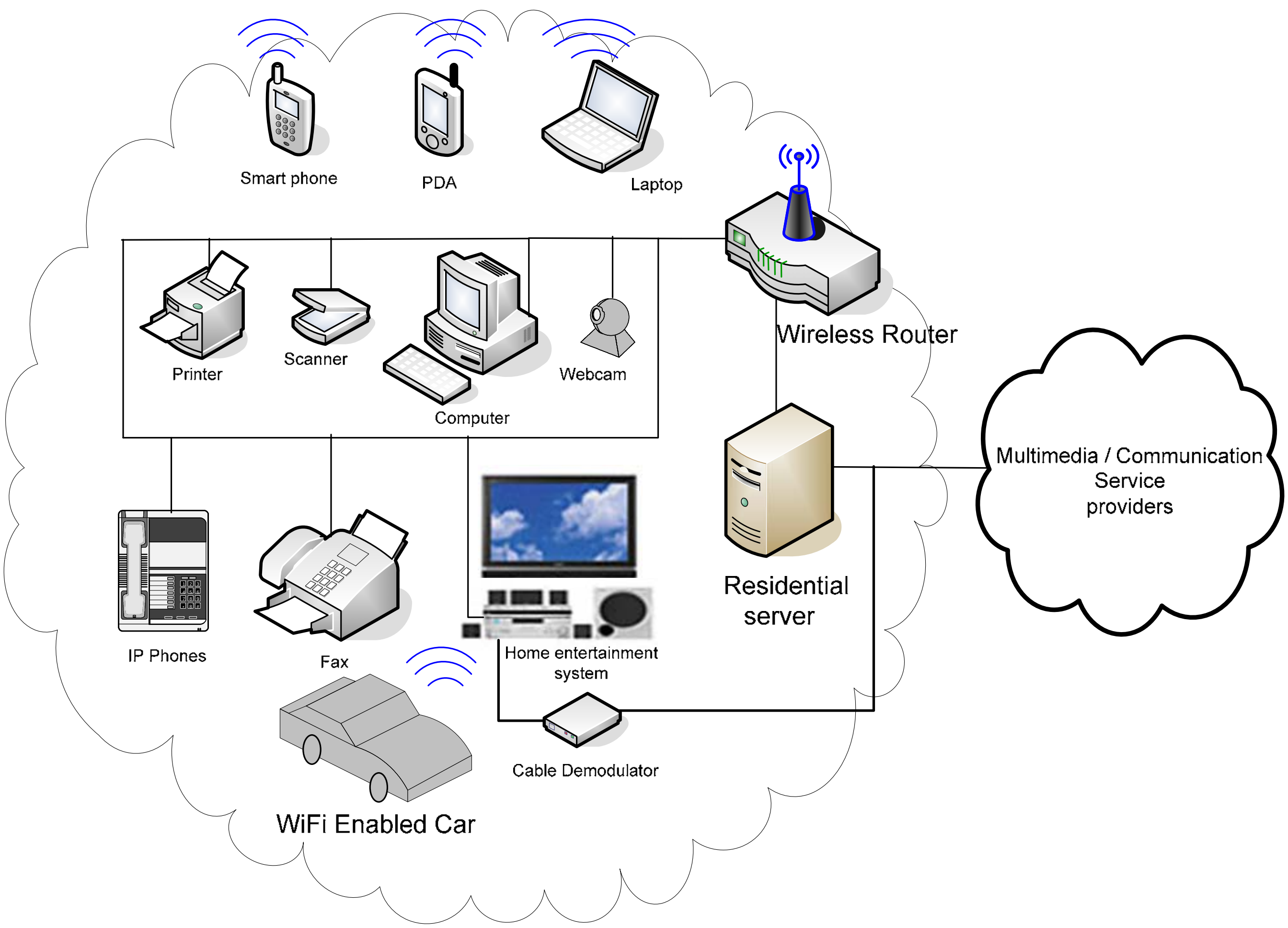

Figure 5.1 Converged Home Network environment

It is obvious that the number of systems and functions can be very large and complicated. The system needs to be backward compatible as well as future proof (future technologies and systems could be integrated without much effort). It would be good idea to start from a very simple system and build on it.

5.2.1 Scenario being considered in Converged residential network

We shall consider a family comprising four people, two parents and two children. Each of these four users might need different services and will have different levels of authority over the system. Let's consider a problem or a convergence requirement of this residential network and work on providing a solution for it.

We assume that say the father (John) has a cell phone (GSM/UMTS). He also subscribes to digital television services from the same company which provides him with a digital telephone and Internet assess. This is a typical scenario at present for a customer of triple play services.

What John desires from this converged system is to have the following features:

- When at home, he should be able to make and receive calls (voice and maybe video) using his home digital phone or cell phone interchangeably. This means that once he gets home his cell phone calls should be automatically routed to his home phone to save minutes.
- The home phone is connected to a high bandwidth connection; it's an IP phone and should be allowed to be implemented over the WiFi network. The cell phone used by John may be a smart phone which has WiFi connectivity, so he can continue to use his handset to receive calls intended for both his home phone number and cell phone number but with enhanced bandwidth, improved display and reduced cost of access.
- Other members of the house may or may not have a cell phone account; if they do same should apply for them. Also the system has to be smart enough to identify the called party and not alert John on his phone if the call is for his children. This

needs to work both ways as he does not need to his calls to be forwarded to other members of his family. The system should be configurable to manage this.

- He should be able to continue to access the internet from any other device, including a desktop, a laptop, a PDA/Smart phone etc. Also he should be able to receive communication (voice call, email, voice mail, instant message, video call, etc.) addressed to his various accounts (email, home phone number, cell phone number, etc.) on any of the devices listed above. They have different abilities in terms of processing power, screen size, etc.
- The entertainment services being subscribed by the family may include digital TV, access to other online multimedia services, music, videos etc. The family must have access to the entertainment services from any network node capable of playing audio and video. This includes watching a particular TV channel on a PC, a laptop or a PDA.
- The family might like to share some data among themselves and their friends; this might be photos, video, audio or other data. They should be able to do that securely, from within the home and even outside home. John might use a database server which would be accessible to all authorized people.
- John would like to be able to add services or applications being run in the home network without changing the system much and should be able to remotely configure and control those services. These might include a security system, a web server, etc.

The requirements mentioned above are quite advanced and may not be implemented any time soon, but the NGN system that needs to be designed should be able to handle all these and more. In the following sections I will propose a means of establishing such a system based on the enhanced IMS architecture.

5.3 Achieving a converged residential network using IMS

The requirements mentioned in the previous section can be achieved by using IMS; it would require some modifications to the architecture but that is not an issue as IMS is a flexible system and can be modified or extended as per needs of the application.

It would not be possible to examine all the aspects of the system, as it is too vast. I will introduce the main concepts in the architecture of the system that would be in accordance with the IMS architecture and its capabilities. There are a few prerequisites prior to establishing sessions in the IMS environment; they are presented in some detail in the following section.

5.3.1 Prerequisites for operation in IMS

There are a few prerequisites that need to be met before the user can operate in IMS. The steps to be followed are as mentioned below:

5.3.1.1 Establishing an IMS service contract

This includes establishing a subscription with the IMS service provider. During this process, the service provider will provide the customer with the appropriate identities and the service profiles will be created depending on the users requirements in terms of services, bandwidth for those services, and access to various other applications being provided by the service provider. We need to keep in mind that we are talking about a general IMS service contract and it is no way limited to the residential environment alone. It definitely would include a wireless cellular subscription, as IMS was designed specifically for the wireless medium.

After the service contract is established the user profile will be stored in the HSS and will be used during various operations in IMS, including Authentication, Authorization and Accounting Purposes, etc. As mentioned in the previous chapter the user profile in IMS may include multiple service profiles, which may contain more than one public identification and initial filter criterion. Filter criterion is defined as set of user related information, which allows the S-CSCF to make a decision about invoking the services of an Application Server (e.g., forward a SIP request). More information about the filter criterion will be provided when we talk about the application servers. Figure 5.2 provides a simple depiction of the contents of a user profile for a particular user; as shown in the figure, a User Profile is attached to a Private User Identity which in turn is attached to number of Public User Identities. They are grouped into a number of Service Profiles. A Service Profile defines the services and the triggers associated with a set up Public User Identities in the form of one or more filter criterion. It also contains an optional entity called the Core Network Service Authorization, which contains a Subscribed Media Profile Identifier. This identifier contains a value which defines the set of SDP (Session Description Protocol) parameters the user is allowed to request. This allows the service provider to define the limitations on the user in terms of the resources it can request.

The public identification contains either a SIP URI (Uniform Resource Identifier) or a TEL URI. All the public identities mentioned under a Service Profile fall under the influence of all the policies enforced by that service profile. Each public identification contains a tag which identifies if it’s barred. A barred Public User Profile will be used only for registration purposes but not for any SIP traffic.

5.3.1.2 Obtaining an IP address

Every IMS terminal needs to get connected to the IMS core network. The connectivity is provided by the IP-CAN (IP –Connectivity Access Network). This could be any IP based transport network such as GPRS (as in GSM/UMTS network), xDSL (Digital Subscriber Line, provided by various internet providers), Wireless LAN through WiFi (IEEE 802.11) or WiMax (IEEE 802.16) networks etc.

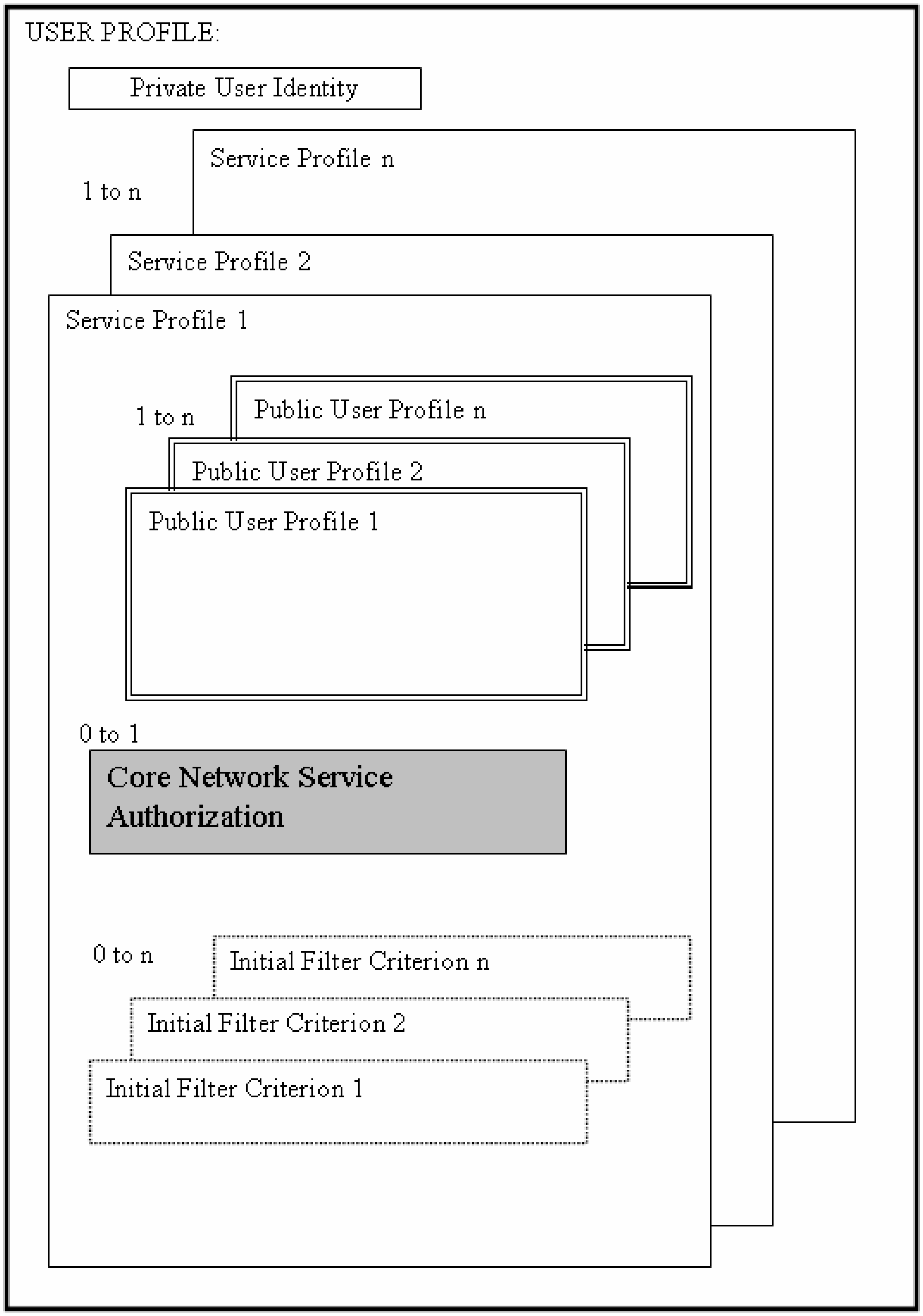

Figure 5.2 User Profile, Organization and Structure in IMS

All these networks dynamically allocate an IP address for a specified period of time and they use different mechanisms for the same. But there is a condition enforced on all IMS terminals; they work exclusively with IP version 6 (IPv6 RFC 2460 [98]). The procedure for acquiring an IPv6 IP address using GPRS is defined in 3GPP TS 23.060 [99]. Other systems are most likely to use DHCPv6 (Dynamic Host Configuration Protocol for IPv6) defined in RFC 3315 [100]. This protocol is used to configure the IMS terminals and provide them with a suitable IPv6 address; this is done as mentioned by using DHCPv6 protocol between the IMS terminal and the DHCP server. This protocol also allows the IMS terminal to receive other data such as the address of an outbound SIP proxy (local CSCF). This is explained in the following paragraph.

5.3.1.3 Discovery of P-CSCF

After the IMS terminal obtains in IPv6 address, the next step is to locate a P-CSCF. This procedure includes the discovery of the IP address of the P-CSCF, which acts as an inbound/ outbound SIP proxy sever and will be present in all the interactions with the IMS core network.

This operation can be done in two ways.

- The P-CSCF discovery procedure could be integrated with the process of obtaining the IP address.
- The P-CSCF discovery could be done as a stand alone process after the IPv6 address has been obtained.

The first method is used in the GPRS environment; once the GPRS attach procedure is completed, it performs a procedure which is called ‘Activate PDP Context Procedure’, defined in 3GPP TS 23.060 [99]. PDP stands for Packet Data Protocol; Internet Protocol is an example of PDP. The completion of this procedure provides the IMS terminal an IPv6 address as well as the IP address of a proxy SIP server.

The second method, called the standalone method, is based in DHCPv6 protocol and DNS (Domain Name System) protocol. After the IMS terminal obtains an IPv6 address, it sends a request to the DHCP server on the reserved multicast address. This message contains a request for the DHCPv6 Options for SIP servers. This process is specified in RFC 3319 [101]. There are two options available to the IMS terminal. The first one involves the DNS server as well as the DHCP server; in this case the IMS terminal requests a list of P-CSCF domain names and the IP address of one or more DNS servers. The DHCP responds with the same, the IMS terminal resolves the domain name of a P-CSCF using one of the available DNS servers and obtains the IP address of the P-CSCF.

In the second option the terminal requests the DHCP for a list of SIP server IPv6 addresses. The response from the DHCP includes the same and the terminal obtains the IP address of the allotted P-CSCF.

5.3.1.4 IMS Registration

The registration process in IMS is based on the SIP registration process, where a Public User Identity is bound to a SIP URI. This SIP URI contains the IPv6 address or the host name of the terminal where the user is reachable. This is done using the SIP REGISTER request, defined in RFC 3261 [57]. The SIP protocol is based on HTTP (Hypertext Transfer Protocol) defined in the RFC 2616 [102]. It is a textual request response protocol. The format of a SIP Message is shown below.

```
Start line
A number of header fields
Empty line
Optional message body
```

The format of a SIP message is described in detail in chapter four of [54]. We will not go into details of the SIP protocol as it is very extensive.

In this section attention will be paid to the functions performed during the registration process rather than how they are implemented using the SIP protocol.

The IMS registration process is heavily overloaded as compared to the normal SIP based registration in the internet world. This is done due to fulfill the 3GPP requirements in a minimum number of round trips. In IMS the goal is complete the registration process in two roundtrips. This is depicted in the following sub section. As mentioned earlier we will assume that the users are IMS subscribers and the have been provided with an ISIM.

5.3.1.4.1 IMS registration using an ISIM

The subscriber terminal needs to have access to a UICC, which contains the ISIM application, the USIM application or both. In this case we will are considering the ISIM application is present and contains all the information which would be required to register and setup sessions.

The initial SIP REGISTER request is created by the IMS terminal after it retrieves the following from the ISIM application:

- **The Private user identity**: used to authenticate the user, it is not used for routing of the messages
- **The Public user Identity**: used to represent the user ID being registered
- **The registration URI**: it's the SIP URI that is used to denote the home network domain

The registration process in IMS needs to accomplish the following tasks:

- Bind a Public User Identity to a contact address (IPv6 address or a host name of the terminal)
- Authentication of the user by the home network
- Authentication of the network by the user
- The SIP registration is authorized by the home network and allows for the usage of the IMS resources subscribed by the user
- Verification of a roaming agreement between the home network and the visited network if the P-CSCF is located outside the home network, thus authorizing the usage of resources
- The home network informing the user about the various other identities that have been allocated to the user (implicitly registered user identities)
- Negotiation of the security mechanisms between the IMS terminal and the P-CSCF for subsequent signaling and other such security associations to protect the integrity of the messages
- Uploading of the compression algorithms between the IMS terminal and the P-CSCF.

These tasks are accomplished in two roundtrips. The registration flow is depicted in figure 5.3 and is briefly described below. In this case we are assuming a simple IMS network where there is just one HSS. In such a case there is no need for an SLF to be present or invoked during the registration process.

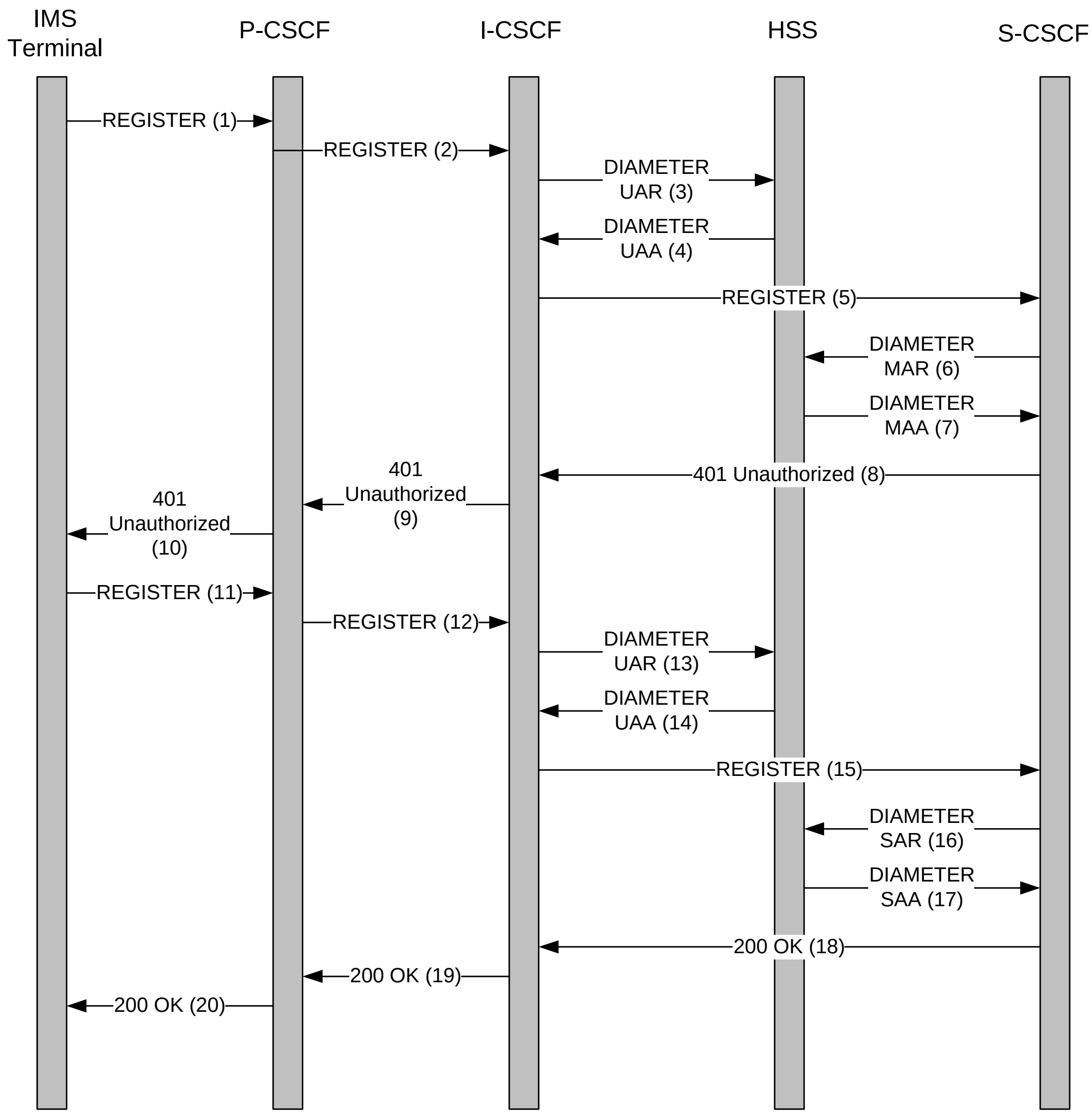

Figure 5.3 Registration flow in IMS

The registration process includes two round trips; step 1 through 10 represent the first round trip and steps 11 through 20 represent the second round trip. The steps performed are as below

Step1: The IMS terminal creates a SIP REGISTER request and includes all the necessary information required to register the Public User Identity as obtained from the ISIM application. The format used is the SIP message format mentioned earlier. The content of this SIP REGISTER request also includes the security association establishment information. IMS requires the establishment of two IPsec security associations between the IMS terminal and the P-CSCF as specified in RFC 3329 [103]. This is done by the terminal by adding a `security-client` header in the REGISTER request which contains the mechanisms (ipsec-3gpp), the algorithm and the SPIs (Identifiers for Security Associations) and the relevant ports to be used.

This SIP message is forwarded to the allocated P-CSCF.

Step 2: Once the P-CSCF receives the REGISTER request it performs certain operations on the message as per its duties mentioned in chapter 4. The P-CSCF extracts and removes the security association related information from the message and adds some more information which includes the network information where the P-CSCF is located; it adds that in `P-Visited-Network-ID` and also adds its own SIP URI in the `Path` header of the SIP REGISTER request. As the P-CSCF may or may not be located in the same network, it needs an entry point to the home network, which is obtained by performing a DNS request as per RFC 3263 [65]. Thus the P-CSCF obtains a valid SIP

URI of an I-CSCF for the home network and the SIP REGISTER request is forwarded to it.

Steps 3 and 4: The I-CSCF receives the Register request from the P-CSCF allocated to the IMS terminal. The I-CSCF does not keep state information for a particular request as a different one may be allocated to the P-CSCF the next time it sends a SIP message. This is due to the load balancing mechanisms implemented by the DNS. The I-CSCF is not aware if an S-CSCF is allocated to a particular user or its address. First step authorization is carried out by the I-CSCF as it sends a DIAMETER User-Authentication-Request (UAR) to the HSS (step 3). This request includes the public and private user identities of the user and the visited network identifier, all obtained form the REGISTER request.
The HSS receives this information and authorizes the user to roam a particular network and authenticates that the Private User Identity has a listed Public User Identity being registered.
Step 4: In this process is the HSS sending a DIAMETER User-Authentication-Answer (UAA) message to the I-CSCF, which conveys to the I-CSCF that the visited network and the User identity is valid. It also includes in this message the SIP URI of a previously allocates S-CSCF to the user. This is in case the user has been previously registered. On the other hand if this is the first time the user is being registered (in case of terminal being powered on) the HSS sends to the I-CSCF a set of capabilities expected from an S-CSCF for that user. This allows the I-CSCF to allocate an S-CSCF that matches the needs of a particular user.

Step 5: The S-CSCF capabilities received by the I-CSCF from the HSS for a particular user are divided into mandatory capabilities and the optional capabilities. They are identified by integers; there are no standards for defining and identifying these abilities, it completely depends on the home network operator. The I-CSCF maintains a list of all the S-CSCF available and their abilities and chooses an appropriate one if one is not already allocated and communicated in the UAA message from the HSS. Step 5 concludes when the I-CSCF proxies the SIP REGISTER request to the selected S-CSCF.

Step 6 and 7: The allocated S-CSCF receives the REGISTER request and extracts all the relevant information about the user being registered. The S-CSCF needs to authenticate the user, which needs to be done only during registration; the S-CSCF does not authenticate the user during other SIP requests such as INVITE, etc. In Step 6 the S-CSCF contacts the HSS and sends a DIAMETER Multimedia-Authorization-Request (MAR). In this request the S-CSCF queries the HSS for the authentication information of the user and at the same time informs the HSS of its address to be stored and provided for future register requests for that user.

In step 7 the HSS answers the S-CSCF with a Multimedia-Authorization-Answer (MAA) which includes the authentication data for that user. The authentication data is referred to as authentication vectors; these authentication vectors, which may be one or more in number, are used by the S-CSCF to authenticate the user.

Step 8 and 9: After the S-CSCF receives the authentication vectors, it creates a response to the REGISTER request; it's denoted as SIP 401 (Unauthorized) response, in which the

S-CSCF includes a challenge in the `WWW-Authenticate` header. This is forwarded to the I-CSCF and the P-CSCF in steps 8 and 9 as per the SIP procedures.

Step 10: The P-CSCF receives the SIP 401 Unauthorized message from the I-CSCF and inserts the security association information in the `Security-Server` header. This modified SIP 401 (Unauthorized) message is forwarded to the IMS terminal in step 10. This completes the security association procedure between the IMS terminal and the P-CSCF. The terminal receives a challenge and will respond by generating another REGISTER request.

Step 11: The challenge by the S-CSCF is a means of authenticating the IMS user. The response to this challenge also known as credentials is generated by the IMS terminal. In this case the authentication parameter is available in the ISIM application present on the smart card (UICC). It is extracted and a response to the challenge is generated by the terminal it is included in a new REGISTER request generated by the terminal and is sent to the P-CSCF as shown in step 11. This new REGISTER request also includes a `security-verify` header which is used verify the security association between the terminal and the P-CSCF.

Step 12: Once the P-CSCF receives this new REGISTER request is performs the same tasks as in step 2, it queries for an I-CSCF and forwards the new REGISTER request after extracting and removing the security association header.

Step 13, 14 and 15: The I-CSCF receives the new REGISTER request and performs the same steps as earlier in steps 3, 4 and 5; except that this time the HSS will answer the UAR with a UAA which contains the address of the allocated S-CSCF. The I-CSCF forwards the new REGISTER request to the allocated S-CSCF which contains the user credentials.

Steps 16 and 17: The S-CSCF will receive the new REGISTER request and extract the user credentials from it, and perform the validation of the received credentials against the ones obtained earlier from the HSS in step 7. If the user is correctly authenticated the S-CSCF sends the HSS a DIAMETER Server Assignment Request (SAR) message to convey the news that the user is now authenticated, this is done in step 16. The S-CFCF receives a response from the HSS in the form of a DIAMETER Server Assignment Answer (SAA) in step 17, which includes the user profile. The contents of the user profile were described earlier and included the initial filer criterion and the implicitly registered public user identities. The S-CSCF performs a number of operations at this stage as per the user profile. It also stores the contact URI of the public user identity and also stores the list of URIs present in the `path` header, which always includes the P-CSCF but may or may not include the I-CSCF; these URIs are used to route any messages intended for that user.

Steps 18, 19 and 20: These are the last steps in the registration process. The S-CSCF generates the SIP 200 (OK) response to the REGISTER request indicating its success. This response contains some additional information, a header called `service-route` which contains a list of SIP server URIs; this list always includes the S-CSCF. Also there is a `P-Associated-URI` header field which includes a list of additional URIs allocated or reserved for the user by the network operator. The SIP 200 OK response is forwarded to the same I-CSCF (step 18) and to the P-CSCF (step 19) and finally reaches the IMS terminal (step 20). This final step completes the process of registration of a particular Public User Identity.

After these prerequisites are accomplished the IMS terminal can establish and receive multimedia sessions, as per the SIP procedures defined in RFC 3261 [57] and IMS resources available to the user. We will not discuss the establishment of these multimedia sessions but assume their working. The components of IMS described so far allow for these multimedia sessions to be established and controlled.

5.3.2 Basic assumptions for the converged architecture

The converged architecture in the residential environment is being achieved using IMS, this means that the users will be IMS subscribers and will be assigned Public and Private User Identities by the IMS provider.

The IMS provider may assign two of more Public User Identities. In this case, John might require more than two, as he wishes to keep his personal contact hidden from the people at work. Same would apply for the other members of the family. The IMS provider will provide the users with a UICC (Universal Integrated Circuit Card), this may contain a ISIM (IMS SIM) application, or an USIM (UMTS SIM) application or both. The information is stored on this card includes among other things the Private User Identity, and one or more Public User Identities. Using this information the user can establish IP Multimedia Sessions in a variety of ways using the IMS architecture. But before we proceed there is need for making a few assumptions or establishing a few conditions on the residential network. They are as below:

- The IP Connectivity Assess Network (IP-CAN) used by the residential is operated by the IMS provider. This is not essential, but makes this system less complicated and easier to understand.
- The residential network is an IP based Ethernet environment, and more specifically uses IPv6 Protocol.
- All the networked devices have IPv6 interfaces (NIC) and can obtain a globally unique IPv6 address.
- We assume that there are no traditional PSTN terminals being used and all terminals are IP based; but the IMS terminals and users are fully capable of calling PSTN or other CS customers and vice versa.
- The entertainment services are digital and use configurable IP based set top devices.

With the above mentioned assumptions in place, we can establish an IMS based network in the residential environment.

5.3.3 IMS implementation for this scenario

Due to the flexibility offered by IMS there can be more than one way in which this system can be setup and implemented. In this section I will provide a solution which in my opinion would be most efficient and would allow for further enhancement of the system.

5.3.3.1 User Profile

For the sake of simplicity let us consider a single user (John) being served by the IMS provider, who has four public user identities; two of which are meant to be used as his general contact info for the public. The other two identities are meant for more personal purposes, and to be used by family and friends. Each of these would include a SIP URI and a TEL URI.

Johns profile is shown in figure 5.4 below

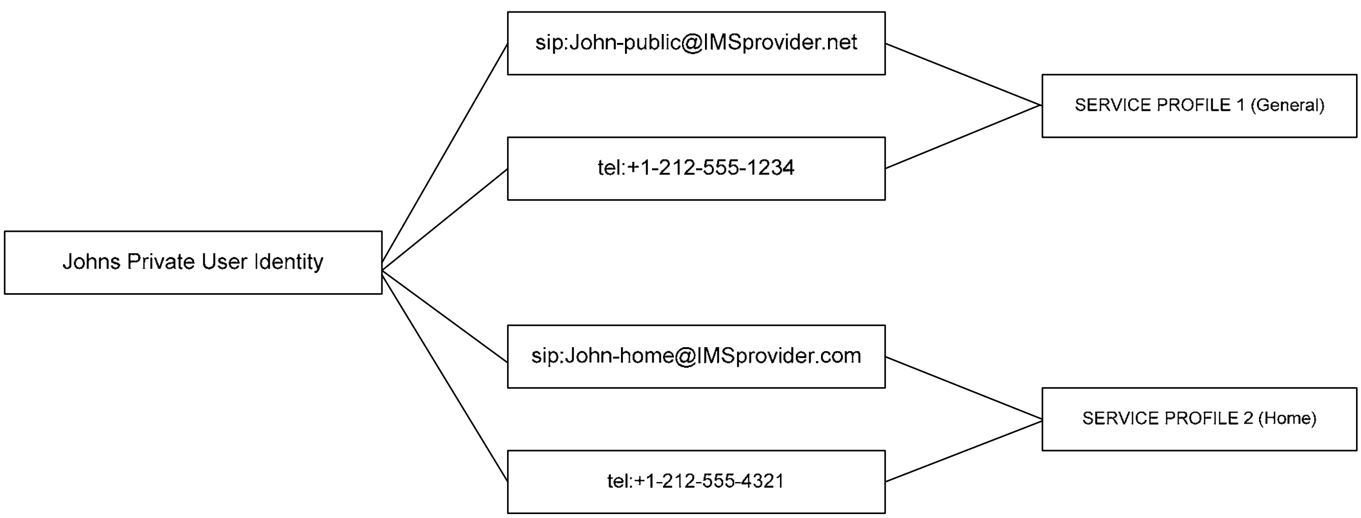

Figure 5.4 Johns User Profile

One of the TEL identities can be thought of as his cell phone number (under the general service profile) and residential phone number (under the Home service profile). There is a SIP URI attached to both these TEL URIs. When John is registered under the John-public identity he will be able receive (voice, video, and other multimedia calls) addressed to either the SIP URI or the TEL URI. This is so because when he registers using the John-public@IMSprovider.net identity his TEL identity is implicitly registered being in the same service profile. The same thing applies for the other set of public identities in the Home Service Profile.

5.3.3.2 Objective for John's Converged home Network and proposed architecture

One of the requirements of a converged residential environment is to allow for his cell phone calls to be forwarded to the residential line if so desired. This can be achieved with the help of an application server which forwards all the calls destined for the identities in the General Service Profile to the location of the public identities registered under the Home Service Profile. Before we look at the process by which this task is accomplished we should look at the way in which IMS is implemented in the residential environment. Figure 5.5 depicts the proposed architecture. Now in this case we assume that the IMS service provider also provides the IP-CAN (IP- Connectivity Access Network) so all the nodes in the residential environment can be considered to be a part of the Home (IMS providers') network. The location of the P-CSCF and the SIP Application Server is unorthodox in the network shown below. But they technically still are in the home network of the IMS provider. This may not always be the case, if so the P-CSCF and the SIP-AS can be considered to be located in a visited network which is permitted in the IMS architecture.

The point is that this architecture and system allows for a high level of flexibility in terms of the IMS service provider. The service provider can be changed without any major changes in the network, except for some reconfiguration of the nodes.

In our case the IMS service provider also provides the IP-CAN and possibly the entertainment services. We will not discuss the entertainment services in detail.

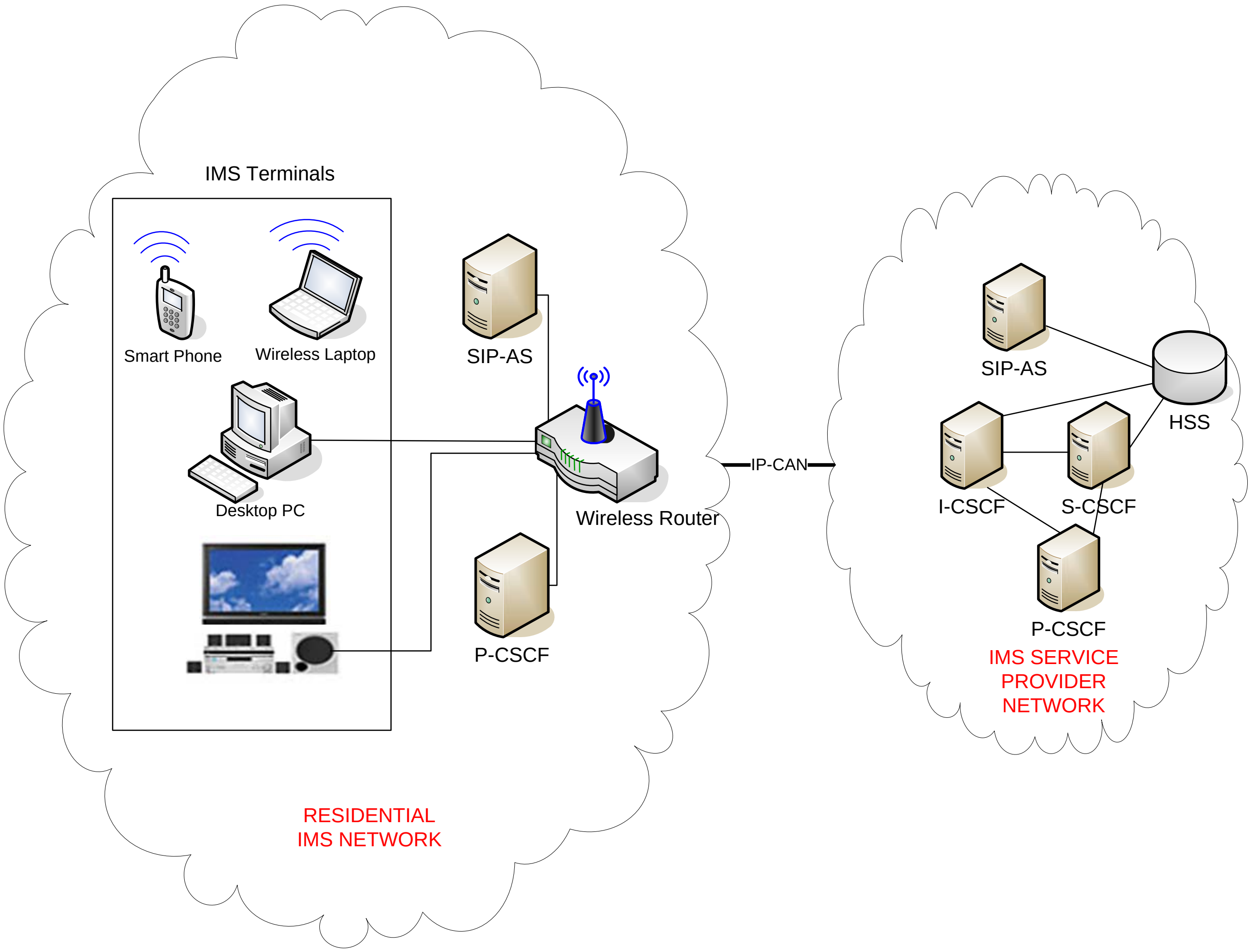

Figure 5.5 Proposed IMS implementation in the Residential environment

5.3.3.3 Implementation of proposed IMS architecture

The IMS agreement is established to allow the user (in this case John) to register any of the public user identities at any of the available IMS terminals. The abilities of the terminal and the subscription of John regulate the type of sessions that can be established.

The IMS terminals obtain an IPv6 address from the IP-CAN provider using the required DHCPv6 and DNS procedures.

The P-CSCF allotment is also regulated by the IMS service provider, in this case we shall assume that the service provider has located an outbound SIP proxy (P-CSCF) at the customer's premises and this P-CSCF will be allocated to any sessions established by the user from the residential network; this may be done by configuring the DHCPv6 for a certain set of addresses or terminal names. This is unusual but has certain advantages mentioned in the following sections.

The registration process followed would be same as described in section 5.3.1. The user information stored in the USIM or ISIM (present in the UICC) would need to be accessed and transmitted in the right format from the IMS terminal. To accomplish this task the IMS terminal being used would need to be connected in some way with the smart card allocated the user. This smart card would probably be present in the wireless handheld device (probably a smart phone) being used by the user, and the user might perform this task by docking the smart phone using a cable or running a secure wireless application which transmits the relevant information from the smart card to the IMS terminal used. In other words the IMS terminal and user needs to be authenticated by the network and the only way to do that is to access the authentication information stored on the smart card (UICC) and there has be a secure means of doing the same. Once the authentication and authorization is complete and the S-CSCF is allocated to the user, it goes thorough the process of evaluating the initial filter criterion, which may or may not result in the invocation of a one or more Application Servers. We have discussed Application Servers in some detail in section 4.3.2.5 where the various modes of operation of Application servers were mentioned.

In our case, we need to provide convergence between the wireless cell phone used by John and the residential digital phone present in the residential network. As shown in figure 5.4 there are two service profiles being used by John. Our task at hand when John is registered under the Home service profile is to handle all incoming calls (voice, video or multimedia) to John's general profile as per John's directive. He may wish for calls from certain people to be forwarded to his Home contact address (his residential number) and the rest of them might be forwarded to some sort of automated response and messaging system (answering machine); there are various other possibilities or ways in which John wishes to be reached while he is registered using his home profile.

This is achieved using the initial filter criterion in present in John's user profile. Let us assume that John maintains a list of people (with known contacts) from whom he wishes to establish incoming communication no matter where he is registered (home or outside). This list is stored in one of the application servers (lets call it AS1). Then there another Application Server (lets call it AS2) which receives input from AS1 regarding the session and acts as a SIP proxy and directs the session to the appropriate location. Both these Application Servers can be configured by the user over the 'Ut' interface; this interface is between the User Equipment and the Application Server and is used exclusively for the purpose of configuring access related information and not for live traffic. The security functions for this interface are defined in 3GPP TS 33.222 [104].

When John registers at home the S-CSCF invokes the AS1 which has been configured by John as per his wishes and regulates his presence at home. This information is used as an input to AS2 which receives the contact address of John and AS1. AS2 will also have access to the list of people who would be permitted to contact John at home. These actions are carried out during the registration process; they are shown in figure 5.6 below.

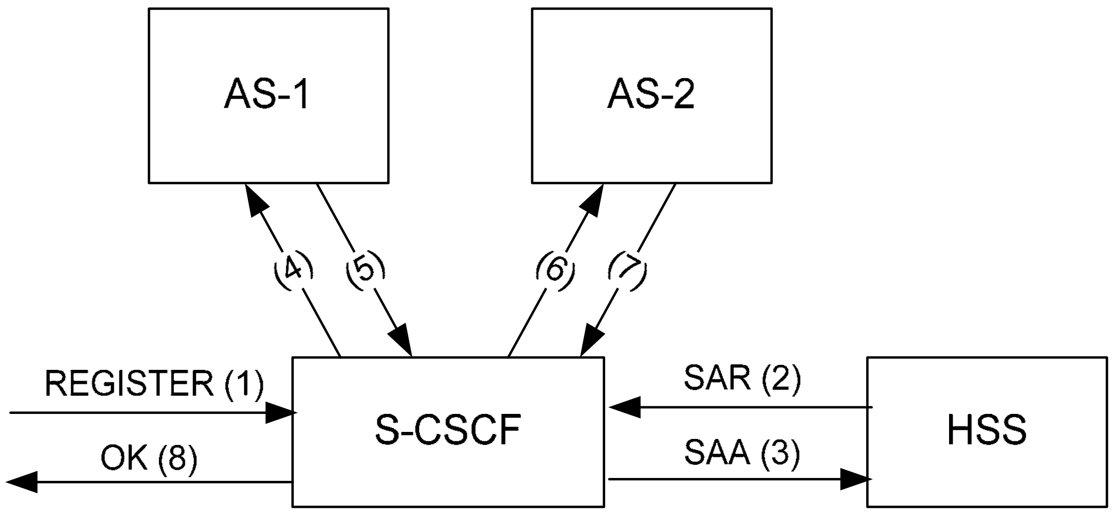

Figure 5.6 Registration and Service Control procedures for John at Home

We will consider steps 1 through 8 to explain the process that takes place. In step 1 the S-CSCF receives the REGISTER request from the I-CSCF (step 15 in the general registration flow diagram figure 5.3). The S-CSCF contacts the HSS and receives a response in steps 2 and 3 (16 and 17 in figure 5.3) and obtains Johns User profile, which includes a set of initial filter criterion. Upon evaluation of the initial filter criterion the S-CSCF invokes AS1 which has been configured by John to reflect his visibility and presence when registered at home.

In this case we assume that he wants to grant access to all users who use his home user identity (telephone or SIP URI) and only a set of users who try to contact him using his general user identities (this might include his boss or other important people). He maintains a list on AS1. The configurations are such that when he registers using his Home profile, he is automatically de-registered by the network from his General Service profile.

In step 4 the S-CSCF will transfer the SIP request to AS1; it modifies the register request to reflect the options chosen by John and sends the request back to the S-CSCF in step 5. At this stage the S-CSCF evaluates the next filter criterion and uses the modified register request, this leads to the invocation of AS2 and the SIP REGISTER request is forwarded to it in step 6. AS2 stores the contact information of the user (present in the request) and also stores the contact info of the Application Server (AS1 in this case), which will be used to access the list of people who are permitted to reach John at home identity. AS2 in step 7 sends the SIP REGISTER request back to the S-CSCF modifying the contents to reflect the tasks performed. AS2 may in turn contact the HSS and modify the contents of the User profile and include the address of AS1 in one of the filter criterion for John's general user profile identities and may also trigger the De-registration process of John's General Profile. In step 8 the S-CSCF sends an OK message to complete the registration process.

After this process is complete John can receive and make IMS calls using his Home user identities and at same time receive calls from selected people (who use his general user identities) on his home user terminal.

Figure 5.7 below shows what happens when a user is calling John on his general user identity, while John is registered at home and the calling user is present in the list of allowed users.

In this case the identity being invited for a session is not registered (John-public@IMSprovider.net) and the HSS will allocate a temporary S-CSCF to evaluate the session control processes (including the initial filter evaluation). In this case the filter criterion has been modified to include the address of AS1 as a part of the registration process. As seen below the INVITE request is received by the temporary S-CSCF in step 1. The filter criterion is evaluated and the INVITE request is forwarded to AS1 in step 2. AS1 evaluates if the request URI is present in the list of allowed users, if so it modifies the INVITE request to include the registered identity of John and sends it back to the Temp S-CSCF in step 3. In step 4 and 5 this new request is forwarded to the appropriate S-CSCF determined by the I-CSCF after a query from the HSS. In step 6 we see that this modified INVITE request is forwarded to the appropriate P-CSCF and the normal session establishment process takes place from that point on. In this case, AS1 is behaving as a B2BUA.

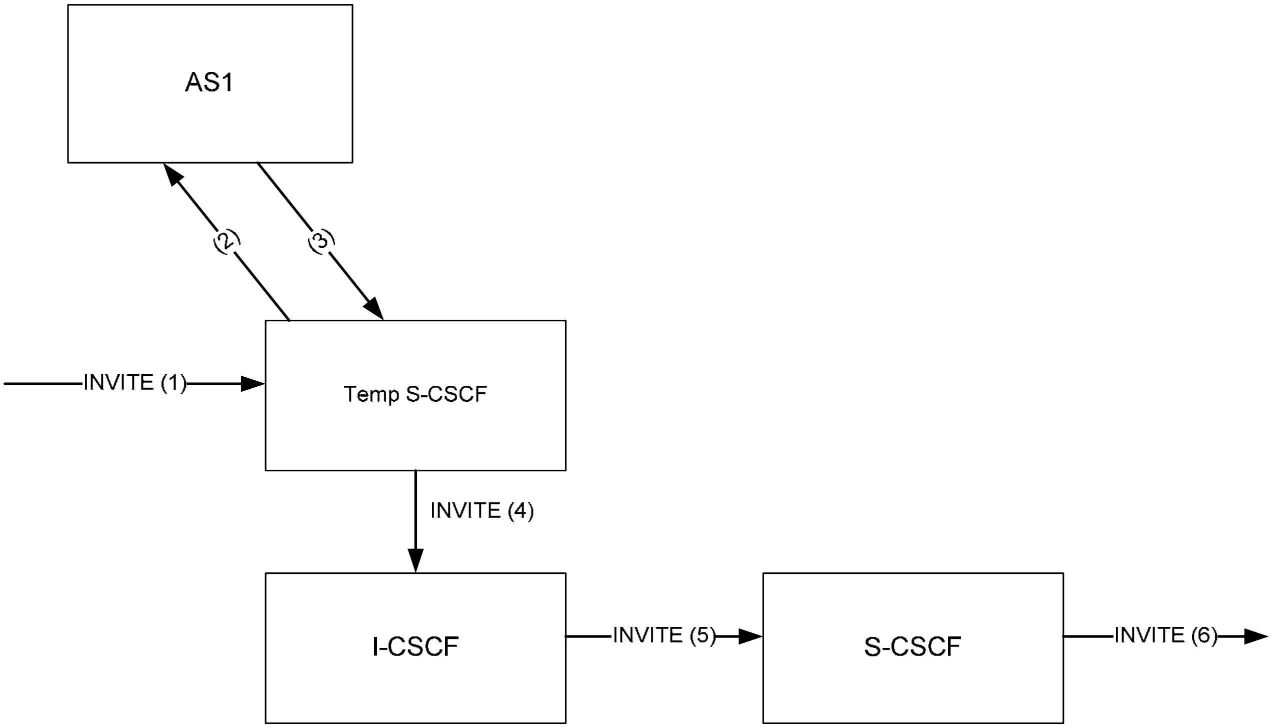

Figure 5.7 Session redirection for users present in John's list

In case the requesting user is not permitted to contact John at home, the session will be transferred to an automated message system which stores a message (like an answering machine but the message could be voice, video or multimedia). This process is shown in figure 5.8 below. It should be noted that for the sake of simplicity we are looking only at the higher levels of signaling used.

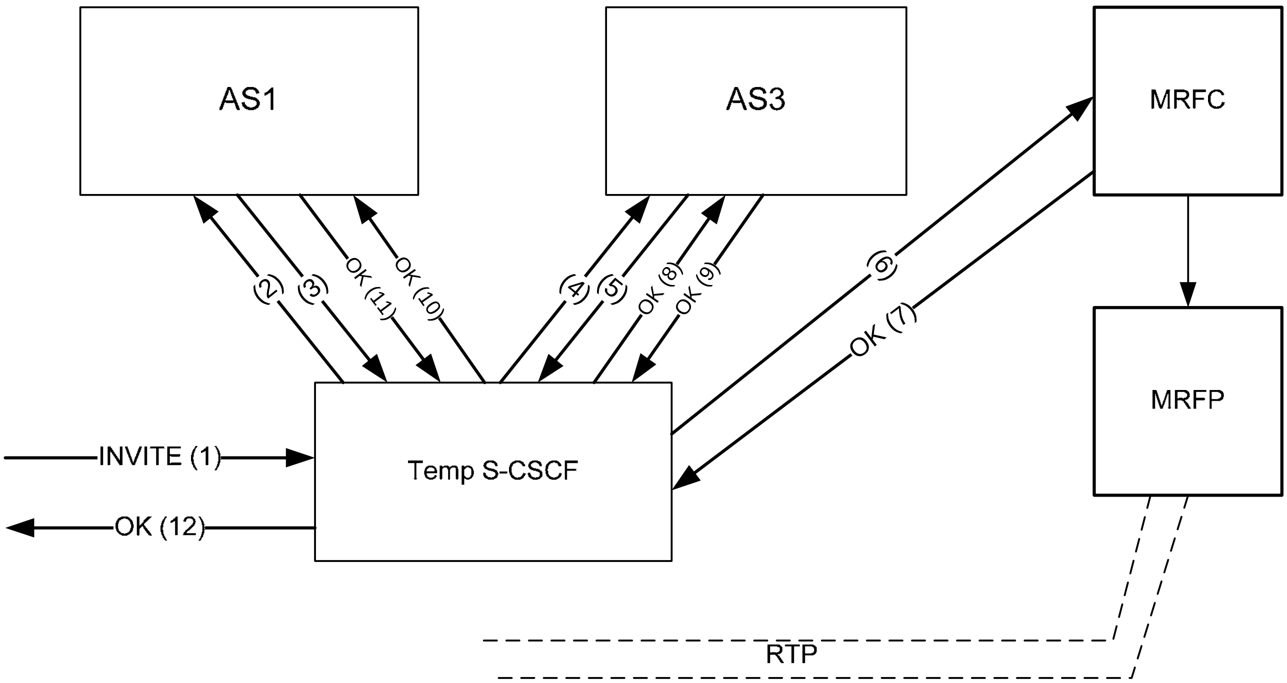

Figure 5.8 Session Redirection to an automated response system for user not in John's list

In this case the first two steps are the same as in figure 5.7. When AS1 receives the INVITE request and finds that the request URI is not in the list it modifies the INVITE request to indicate the same. The INVITE request is sent back to the S-CSCF in step 3. The S-CSCF evaluates the next filter criterion and which results in the invocation of AS3 in step 4 (this does not happen in the previous scenario as the filter criterion evaluation would have failed). AS3 receives the modified invite request in step 4 and changes it further (replaces the Request URI) to include the contact information of the MRFC (Media Resource Function Controller) and sends it back to the S-CSCF; here AS3 is behaving as a SIP proxy. The MRFC controls the playing of the automated response (informing the user that John is unreachable) and recording of the message with the help of the MRFP (Media Resource Function Processor) which uses RTP. The S-CSCF contacts the MRFC in step 6 and receives an OK response in step 7, which is forwarded back to the callee in step 12 after passing through AS1 and AS3.

The procedures mentioned above provide the necessary functionality to offer John a converged network, which fulfills all the AAA, security, QoS, and other such requirements of an advanced NGN network. We have looked at a simple example to automate the redirection of John's calls to the location he is registered, all the different ways of possible redirection of calls is described in section 5.11.5 of 3GPP TS 23.228 [61].

This is just one of the many services that can be provided using the IMS architecture. Similarly many entertainment services can be provided to the users in the form of IP multimedia session and contents of these sessions could be personalized and can be easily managed by the user along with the flexibility of any time anywhere access.

5.3.3.4 Advantages of using IMS in this particular manner

It was mentioned earlier that implementation of IMS in the residential environment as shown in figure 5.5 is not a normal way of implementing IMS. Specifically the placement of the P-CSCF at the residence level is distinguishing. The P-CSCF is generally present in the IMS providers' network or at a visited network which supports IMS. In figure 5.5, I have proposed a system which places a P-CSCF at the customer's premises; this is certainly plausible and does not break any rules of IMS. If the IMS provider is the IP-CAN provider (as considered in our case), the P-CSCF would technically still be a part of the IMS providers' network. If that is not the case and the IP-CAN is not provided by the

IMS provider, the P-CSCF would be considered to be placed in a visited network (IMS enabled) which is supported by IMS. So this system would work in both the scenarios without much modification.

This is one of the main advantages of placing a P-CSCF at the users' premises, but there are other advantages as well. The functionality provided by the P-CSCF is described in chapter 4 (section 4.3.2.3). Placing the P-CSCF at the residence will be beneficial in many ways. They are listed below:

- As mentioned above this system can support both forms of IMS implementation and so would allow a user to use services offered by more than one IMS provider or different members of the family can have different providers or visitors will be able to use their IMS terminals.
- The P-CSCF performs the task of security association establishment with the user equipment. If the user is present in a secure residential environment this process can be ignored as integrity protection would not be of prime concern and even when performed will be much faster if the P-CSCF is located in the same network as the devices.
- The P-CSCF authenticates the user to the rest of the network. As the P-CSCF would be dedicated to certain nodes (fixed devices in the residential network with fixed IPv6 Address), it would make the process less time consuming.
- The P-CSCF discovery procure would be more straightforward and can be statically defined for devices within the IP address range of the residential network. The DHCP can be configured to do this along with the allocation of IP addresses to the various IMS terminals.

- The signal compression and decompression is a task of the P-CSCF, being placed at the residential level, this process would occur at an earlier stage thus reducing the overall session establishment time further.
- The P-CSCF may include a Policy Decision Function (PDF), which regulates the establishment of session based on the resources available and permitted for the user. This information is obtained from the HSS. The establishment and denial of sessions is regulated by this functionality. Being present locally, the decisions can be made faster and would reduce the traffic over the IP-CAN.
- Another advantage of placing a P-CSCF locally is the reduced functionality that needs to be provided by the operator, it is essentially division of work. It would lead to reduced equipment at the operator’s premises, without any loss of control over the system.

This system would certainly be beneficial in bringing the IMS architecture closer to the users and provide the advanced users with additional powerful tools to develop and launch customized service and applications. Figure 5.5 shows an Application Server in the residential network. This is certainly within the rules of IMS as Application Servers can be in the home, visited or a third party network.

The applications may include, but are not limited to, the following:

- A data base server could be implemented at home which is used to store, share and access music, video, photos, and other data among the family members and their friends. This application would provide different levels of access to different users and can be configured and managed remotely by authorized users.

- A multimedia, entertainment/ gaming application could be implemented by an application server which allows authenticated users to view, and record TV shows and movies as per their desire. The user is able to watch the subscribed channels from anywhere (in or outside the home network) as per their convenience. Such a service could be hosted locally at the residence or could be implemented and maintained by the IMS provider.
- Such an application server could be used to automate the electricity, gas, and other utilities used in the home and can be remotely monitored and controlled by the user. This may include a residential security system, which may have multiple sensors and video feeds.

The applications that can be developed based on the IMS architecture are limited only by the imagination of the developer and the needs of the customer. This system combined with the increasing bandwidth capacity available to the user certainly appears to be a winning combination for the first step towards the next generation in communication systems.

5.4 Conclusions

This chapter provides a clear picture of a converged, all IP communications environment, which fulfills almost all the expectations and requirements of a NGN system. The aim was to help the reader comprehend the vision that the telecommunications industry has for the future. IMS is depicted as a major enabler for this vision. IMS was used to implement a specific scenario, which provides a converged network, fulfilling some of the requirements of a user with bundled up communication/entertainment services. The system is not described in detail but sufficiently enough to show that it is achievable, but obviously requires additional work.

Some of the applications mentioned are a bit advanced and not suitable for immediate launch, but are certainly possible to implement. The leaders of the industry are planning to introduce this architecture gradually and in stages; this is a good approach as IMS has yet to prove itself and users need to get used to this new idea of anywhere anytime multimedia communications with a single user identity.

My work in this thesis is focused on understanding these new technologies which will be instrumental in achieving and deploying the NGN systems. I was able to achieve the requirements of a converged residential environment, and was able to propose a network design using the IMS architecture. IMS architecture is a favorite among designers of NGN systems, and almost all the new systems are based on it. Recently Motorola released a residential gateway (at the CES, Las Vegas, Jan 2006) called the Residential Seamless Mobility Gateway (RSG) which achieves Fixed/ Mobile Convergence (FMC). It is based on IMS, which shows the industry is moving ahead with its plans and has started to implement various NGN systems. My thesis has been able to provide me with a better insight, and has allowed me to understand the systems more comprehensively.

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Appendix A – List of abbreviations

3GPP - 3rd Generation Partnership Project
3GPP2 - 3rd Generation Partnership Project 2
AAA - Authentication, Authorization and Accounting protocol
ADSL - Asymmetric DSL
AMC - Adaptive Modulation and Coding
AMPS - Advanced Mobile Phone System
AMR - Adaptive Multirate
AMR-WB - AMR-Wide Band
ANSI - American National Standards Institute
API - Application Programming Interface
ARIB - Association of Radio Industries and Businesses
ARPU - Average Revenue per User
AS - Application Server
ASE - Application Service Elements
ASP - Application Service Providers
ATIS - Alliance for Telecommunications Industry Solutions
ATM - Asynchronous Transfer Mode
B2BUA - SIP Back-to-Back User Agent
BGCF - Breakout Gateway Control Function
BICC - Bearer Independent Call Control
B-ISDN - Broadband Integrated Services Digital Network
BRI - Basic rate interface
BSS Base Station Subsystem
CAMEL - Customized Applications for Mobile network Enhanced Logic
CAP - CAMEL Application Part
CAPEX - capital expense
CAS - Channel-Associated Signaling
CATV - Community Antenna Television
CCS - Common Channel Signaling
CCSA - China Communications Standards Association
CDMA - Code Division Multiple Access
CDMA 2000 - Code Division Multiple Access 2000
CDR - Charging Data Records
CDSL - Consumer DSL
CM - Cable Modem
CMTS - Cable Modem Termination System
COPS - Common Open Policy Service
CS - Circuit Switched

CSCF - Call Session Control Functions
CSMA/CA - Carrier Sense Multiple Access with Collision Avoidance
CSMA/CD - Carrier Sense Multiple Access with Collision Detection
DAMPS - Digital AMPS
DCCP - Datagram Congestion Control Protocol
DLC - Digital Loop Carrier
DNS - Domain Name Service
DOCSIS - Data- Over-Cable Service Interface Specification
DRM - Digital Rights Management
DS - Direct Sequence
DSCP - Differentiated Service CodePoints
DSL - Digital Subscriber Line
DSLAM - Digital Subscriber Line Access Module
DSL-F - DSL Forum
DWDM - Dense WDM
EDGE - Enhanced Data rates for GSM Evolution
ESIF - Emergency Services Interconnection Forum
ESP - Encapsulated Security Payload
ETS - Emergency Telecoms Service
ETSI - European Telecommunications Standards Institute
FC - Fiber Channel
FCC - Fedral Communications Commission
FCC - NRIC - FCC Network Reliability & Interoperability Council
FDD - Frequency Division Duplex
FDDI - Fiber Distributed Data Interface
FDMA - Frequency Division Multiple Access
FM - Frequency Modulation
FR - Frame Relay
GGSN - Gateway GPRS Support Nodes
GPON - Giga Passive Optical Network
GPRS - General Packet Radio Service
GSM - Global System for Mobile communications
HARQ - Hybrid Automatic Request
HDSL - High bit rate Digital Subscriber Line
HFC - Hybrid Optical Fiber Coaxial Cable
HLR - Home Location Register
HSCSD - High-Speed Circuit-Switched Data
HSDPA - High Speed Downlink Packet Access
HSDSL Single-pair High-Speed Digital Subscriber Line
HSS - Home Subscriber Server
HSUPA - High Speed Uplink Packet Access

HTTP - Hypertext Transfer Protocol
I-CSCF - Interrogating CSCF
IDEN - Integrated Dispatch Enhanced Network
IEEE - Institute of Electrical and Electronics Engineers
IETF - Internet Engineering Task Force
IKE - Internet Key Exchange
IMS - Internet Protocol Multimedia Subsystem
IMS-ALG - IMS Application Level Gateway
IMSI - International Mobile Subscriber Identifier
IM-SSF - IP Multimedia Service Switching Function
IN - Intelligent Network
INC - Industry Numbering Committee
IP - Internet Protocol
IP-CAN - IP-Connectivity Access Network
IPv4 - Internet Protocol version 4
IPv6 Internet Protocol version 6
IS-95 - Interim Standard 95
ISC - IP Multimedia Subsystem Service Control
ISDN - Integrated Digital Services Network
ISDN-UP - ISDN-User Part
ISIM - IMS Subscriber Identity Module
ISO - International Organization for Standardization
ITU-T - International Telecommunication Union – Telecommunication Standardization Sector
LAES - Lawfully Authorized Electronic Surveillance
LAN - Local Area Network
LANE - LAN Emulation
LLC - Logical Link Control
LMI - Local Management Interface
MAA - Multimedia-Authorization-Answer
MAC - Media Access Control
MAR - Multimedia-Authorization-Request
MCNS - Multimedia Cable Network System
MEF - Metro Ethernet Forum
MGCF - Media Gateway Control Function
MGCF - Media Gateway Control Function
MGW - Media Gateway
MIMO - Multiple-Input Multiple-Output
MPLS - Multi Protocol Label Switching
MRF - Media Resource Functions
MRFC - Media Resource Function Controller

MRFC - Media Resource Function Controller
MRFP - Media Resource Function Processor
MRP - Market Representation Partners
MSAU - Multi Station Access Unit
MSC - Mobile Switching Center
MSF - Mobile Switching Forum
MTP - Message Transfer Part
NAI - Network Access Identifier
NAMPS - Narrow Band AMPS
NAT-PT - Network Address Translation – Protocol Translation
NENA - National Emergency Numbering Association
NGN - Next Generation Network
NGSP - Next Generation Service Providers
NIC - Network Interface Cards
NMT - Nordic Mobile Telephone
OAMP - Operations Administration Maintenance and Provisioning
OFDM - Orthogonal Frequency-Division Multiplexing
OFDMA - Orthogonal Frequency Division Multiple Access
OMA - Open Mobile Alliance
OPEX - operational expense
OSA - Open Source Access
OSA-SCS - Open Source Access-Service Capability Server
OSI - Open Source Interconnect
PAN - Personal Area Network
PCM - Pulse Code Modulation
PCS - Personal Communication Services
P-CSCF - Proxy CSCF
PDC - Personal Digital Cellular
PDP - Packet Data Protocol
PDP - Policy Decision Point
PEP - Policy Enforcement Point
PHB - Per Hop Behavior
PIB - Policy Information Bases
PLMN - Public Land Mobile Network
POTS - Plain Old Telephone Service
PPP - Point to Point Protocol
PRI - Primary Rate Interface
PS - Packet Switched
PSI - Public Service Identities
PSTN - Public Swithced Telephone Network
RADSL - Rate Adaptive DSL

RFC - Request for Comment
RFID - Radio Frequency Identification
RSVP - Resource reSerVation Protocol
RTP - Real Time Protocol
SAA - Server Assignment Answer
SAR - Server Assignment Request
SCF - Service Switching Function
SCP - Signaling Control Points
S-CSCF – Serving-Call Session Control Function
SCTP - Stream Control Transmission Protocol
SDO - Standard Development Organization
SDP - Session Description Protocol
SDSL - Symmetric Digital Subscriber Line
SEG - Security Gateways
SGSN - Serving GPRS Support Node
SGW - Signaling Gateway
SIM - Subscriber Identification Module
SIP - Session Initiation Protocol
SIP - Session Initiation Protocol
SLF - Subscriber Location Function
SMDS - Switched Multimegabit Data Services
SMR - Specialized Mobile Radio
SMTP - Simple Mail Transfer Protocol
SNMPv3 - Simple Network Management Protocol version 3
SONET - Synchronous Optical Network
SS7 - Common Channel Signaling System 7
SSP - Service Switching Point
SSP - Signaling Switching Points
STP - Signaling Transfer Points
TACS - Total Access Communication Services
TCAP - Transaction Capabilities Application Part
TCP - Transmission Control Protocol
TCP/IP - Transmission Control Protocol/ Internet Protocol
TDMA - Time Division Multiple Access
THIG - Topology Hiding Inter-network Gateway
TIA - Telecommunication Industry Association
TISPAN - Telecoms & Internet converged Services & Protocols for Advanced Networks
TR - Technical Report
TS - Technical Specifications
TSG - Technical Specification Groups

TTA - Telecommunications Technology Association
TTC - Telecommunications Technology Committee
UAA - User-Authentication-Answer
UAR - User-Authentication-Request
UDP - User Datagram Protocol
UE - User Equipment
UICC - Universal Integrated Circuit Card
UMTS - Universal Mobile Telecommunication System
URI - Uniform Resource Identifier
USIM - UMTS Subscriber Identity Module
UTP - Unshielded Twisted Pair
UTRA - Universal Terrestrial Radio Access
VDSL - Very high data rate DSL
VDSL2 - Enhanced VDSL
VLR - Visitor Location Register
WAN - Wide Area Network
WCDMA - Wideband CDMA
WDM - Wavelength Division Multiplexing
WDM-PON - WDM Passive Optical Network
WiFi -WLAN based on IEEE 802.11 a, b, g etc
WiMax - Worldwide Interoperability for Microwave Access
WLAN - Wireless Local Area Network
WPAN - Wireless Personal Area Network
Zigbee - IEEE 802.15.4