Chapter – 1
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In the past few years, the evolution of cellular networks has reflected the success and growth the Internet has experienced in the last decade. This leads to networks where IP connectivity is provided to mobile nodes. The result is third generation (3G) networks where IP services such as voice over IP (VoIP) and instant messaging (IM) are provided to mobile nodes (MN) in addition to connectivity. IP Multimedia Subsystem (IMS) is a new framework, basically specified for mobile networks, for providing Internet Protocol (IP) telecommunication services. It has been introduced by the Third Generation Partnership Project (3GPP) in few phases [1], [2] for Universal Mobile Telecommunications System (UMTS) networks.

The telecommunications world is being revolutionized with the emergence of next generation networking (NGN). Next-generation networks with IP based cores allow convergence of different network architectures in both wired and wireless worlds. The frontrunners of this convergence include SIP (Session Initiation Protocol) and IMS making it possible for triple play services (voice, data and video) to flow over the same network.

The term Next Generation Network (NGN) is used to describe an integrated, open network architecture that provides voice, data and multimedia services over the same network. The concept of integrated services on a single network is not new. Both Broadband ISDN and ATM deliver integrated services. The fact that NGN is a packet-based QoS enabled network providing both telecommunication and data services over different access technologies, sets it apart, making it new and appealing.

Most services today are tightly coupled with a specific transport network and signaling protocol. NGN separates service-related functions from the underlying transport related technologies. This independence between service and transport layer makes the underlying technology invisible to the user regardless of where in a multi-service, multi-protocol, multi-vendor environment the user resides. This concept of nomadicity provides seamless communication between fixed and mobile users. The most important feature of NGN is the converged, QoS aware infrastructure. All information is transmitted as packets that are labeled according to their type (data, voice etc.) and are handled differently based on their QoS criteria. NGN ensures transparent service availability throughout the network by virtualizing services and provisioning seamless access to critical information. NGN provides ubiquitous access in a converged wireless and wireline network by decoupling transport technologies and services.

Many prior works, hands-on done on IMS, Lee and Knight [3] explain the difference between NGN and the Internet since both uses IP as one important protocol. One main difference is that NGN does not restrict service delivery to best effort. NGN is a secure and trustworthy network supporting various services to meet the user's dynamic requirements. It also allows the integration of traditional telephone networks with data communications.

International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) Study Group 13 defined an NGN in Recommendation Year 2001 [4] as:

> *"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 to users by different service providers. It supports generalized mobility, which will allow consistent and ubiquitous provision of services to users."*

Recommendation Y.2001 further characterizes the NGN 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 transport, 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 and multimedia services)
- Broadband capabilities with end-to-end quality of service (QoS)
- 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 and mobile networks
- Independence of service-related functions from underlying transport technologies
- Support of multiple last mile technologies
- Compliance with all regulatory requirements concerning emergency communications, security, privacy etc.

IP Multimedia Subsystem (IMS) is referred as the heart of NGN. IMS is the next generation IP based infrastructure-enabling convergence of data, speech and video and mobile network technology. It is the envisioned solution that will provide new multimedia rich communication services by mixing telecom and data on an access independent IP based architecture, defined in 3rd Generation Partnership Project (3GPP), 3rd Generation Partnership Project 2 (3GPP2) and Internet Engineering Task Force (IETF) standards. The IMS architecture demonstrates in Figure 2.5 used by service providers in NGN networks to offer network controlled multimedia services.

The aim of IMS is to provide all the services, current and future, that the Internet provides with roaming facilities. To achieve these goals, IMS supports peer-to-peer IP communications between existing technology standards while providing a framework for inter-operability of voice and data services for both fixed (POTS, ISDN) and mobile users (802.11, GSM, CDMA, UMTS). It provides session control, connection control and an application services framework with both subscriber and services data, while allowing interoperability of these converged services between subscribers. IMS truly merges the Internet with the cellular world; it uses cellular technologies to provide ubiquitous access and Internet technologies to provide appealing services [20].

IMS essentially replaces the control infrastructure in the traditional circuit-switched telephone network, separating services from the underlying networks that carry them. IMS has a signaling and media plane which work separately, unlike PSTN. The signaling plane handles the session control, authorization, and security and QoS aspects while a media plane manages the media encoding and transport issues. IMS enables services such as text messaging, voice mail and file sharing to reside on application servers anywhere and be delivered by multiple wired and wireless service providers.

1.2 Motivation of Research

The emergence of new technologies has led to the birth of new applications with VoIP, IPTV, and Presence etc. This convergence has increased the demand for services with new performance characteristics; a proposition that decides the future of service provider's business.

Wong and Verma [6] discuss the advantages and disadvantages of providing new IP multimedia service capabilities as opposed to basic IP connectivity from a subscriber, network operator and third party application vendor's perspective. The basic IP connectivity provided by plain vanilla Internet service provider compels the subscribers to use third-party providers for IP multimedia services and applications.

While the network operators may lose potential revenue by not providing basic IP multimedia services like voice-over-IP call capability, it would be difficult to predict which services would be profitable and popular with the customers. So the network operator provides flexibility by allowing third party applications while focusing on their core competency of providing connectivity. A drawback would be lack of reliability, quality issues caused by careless reuse by subscribers of IP multimedia applications, leading to loss of customers and revenue.

Today both cable television and telecommunication operators offer most of communication & entertainment related services. Service providers are struggling with the opportunities and challenges that the convergence presents while striving to gain a foothold in the market segment. It is assumed that an integrated solution with a combination of services will bolster customer satisfaction to encourage loyalty and discourage churn. Interoperability between different service providers is not considered as a design target.

As service providers build out their 3G, broadband and converged networks, they are moving to a business model that makes the quick introduction of new services while exploiting the full revenue potential of these new services and having the flexibility of scaling easily. Service providers must now focus on management of the network resources to optimize the performance of the services they deliver. Expectations of the sophisticated customers from the service providers have multiplied with the new technologies, requiring complicated and customized solutions for their satisfaction. Service providers need to monitor and manage network traffic to optimize performance in order to meet the guaranteed levels of service promised to their customers.

Effective monitoring of network performance and analysis of network data to correlate end-to-end service performance is essential for the thorough understanding of network behavior. Deploying new applications in a service provider environment creates rigorous demands on the network infrastructure and the network operations staff. Service providers are often targets of easy purchase of new technology products from application vendors. Scalability, flexibility and integration of diverse applications is critical to support large and complex networks while reducing the operational cost of the network. The challenges in offering these new services in light of today's highly competitive environment with increased customer focus is, not only seamless transition between disparate network topologies but also the flexibility to embrace new service deployment technologies such as the IP Media Subsystem.

Such issues are mostly associated with determining the right business model, back-end support and economic environment rather than technology. Service providers need a model that makes quick introduction of new services of primary importance, while fully exploiting their full revenue potential and having the flexibility to accommodate easy scaling of networks. For example, using the right billing platform to address a variety of subscriber demographics or having the appropriate subscriber density to financially justify the introduction of a new service are a few factors that affect decisions to offer IP multimedia services.

1.3 Mobile Communication Technologies at Glance

Telecommunications has a long history. The computer age is also half a century old and has now become the Internet age. While their development was largely independent at first, computing and communications have become inextricably bound and mutually dependent.

The Communication world seeks a unified understanding of convergent information and telecommunications services and the underlying network and software technologies.

To visualize the future, need to understand the historical development leading to present day and emerging technologies.

Table 1.1 Timeline for Telecommunication innovation

Duration	Innovation in Telecommunication world
Mid-1970s	• Analog to Digital conversion
1974–1983	• Packet Switching with X.25 standard • TCP/IP adopted by ARPANET predecessor of Internet • N-ISDN • First Optic Fiber Cable Deployed • First Generation Mobile (analogue) network started operation
1984–1993	• Packet switching standards expanded to include Asynchronous Transfer Mode (ATM) and Frame Relay • Launch of World Wide Web, Internet user grew to 1 million • GSM (2nd Generation mobile networks) standardized and successfully launched • Concept of Intelligent Network Deployed in PSTN • Implementation of Value Added Services (VAS) in PSTN • With Synchronous Digital Hierarchy, Network Operator able to configure & provide transmission Services
1994–2003	• Commercialization Internet service providers from Government • Standards for telephony using Internet Protocol (IP) networks • Concept of a New Multiservice Network formulated as the Next Generation Network • The first third Generation (3G) mobile network licences & deployment • Dot.com boom with unlimited optimism about new Internet based services, (Internet User crosses over 100 million) • Optic fiber transmission capacity increased due to both higher speeds of transmission and the use of multiple wavelengths on a single fiber. • Implementation of Interworking between circuit-switched and packet networks to provide the development of media and signalling gateways

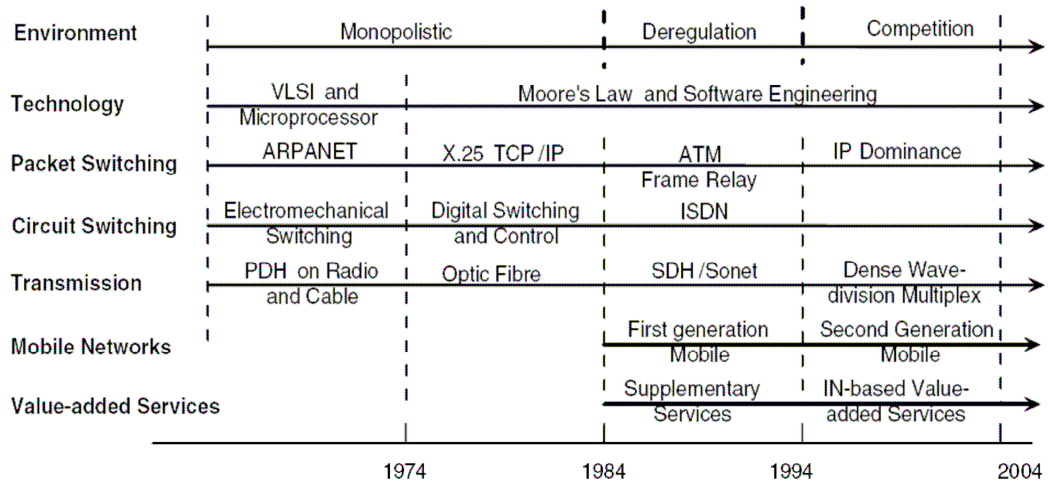

Figure-1.1: Timeline of developments leading to present state of Mobile Communication

1.4 Introduction to Mobile Networks

Wireless telephone comes in two basic varieties: cordless and mobile phones. Cordless phones are devices consisting of a basic station and a handset sold as a set for use within the home.

Mobile phones have gone through three generations, with different technologies:

1. Analog voice.
2. Digital voice.
3. Digital voice and data.

1.4.1 First Generation Mobile Network (Analog voice)

In the earliest mobile radiotelephone (1946), with large transmitter on top of tall building had a single channel, used for sending and receiving. User had to push a button that enables the transmitter and disables the receiver. Improved Mobile Telephone System (IMTS) the push-to-talk (PTT) button was no longer needed. IMTS supported 23 channels spread out from 150 MHz to 450 MHz. Due to the small number of channels, users often had to wait a long time before getting a dial tone [7].

1.4.1.1 Advanced Mobile Phone System.

Advanced Mobile Phone System (AMPS) invented by Bell Labs in 1982. It was also used in England, called TACS, NMT in France. This system is no longer in use, but some of its fundamental properties have been directly inherited by its digital successor D-AMPS, in order to achieve backward compatibility.

In all mobile phone systems, a geographical region is divided up into cells see Figure 1.2. In AMPS, the cells are typically 10 to 20 km across; in digital systems, the cells are smaller. Each cell uses some set of frequencies not used by any of its neighbors. The key idea that gives cellular systems far more capacity than previous systems is the use of relatively small cells and the reuse of transmission frequencies in nearby cells. AMPS system might have 100 cells in 10-km area and be able to have 10 to 15 cells on each frequency. Thus, the cellular design increases the system capacity by at least an order of magnitude, more as the cells get smaller. Smaller cells mean that less power is needed, which leads to smaller and cheaper transmitters and handsets.

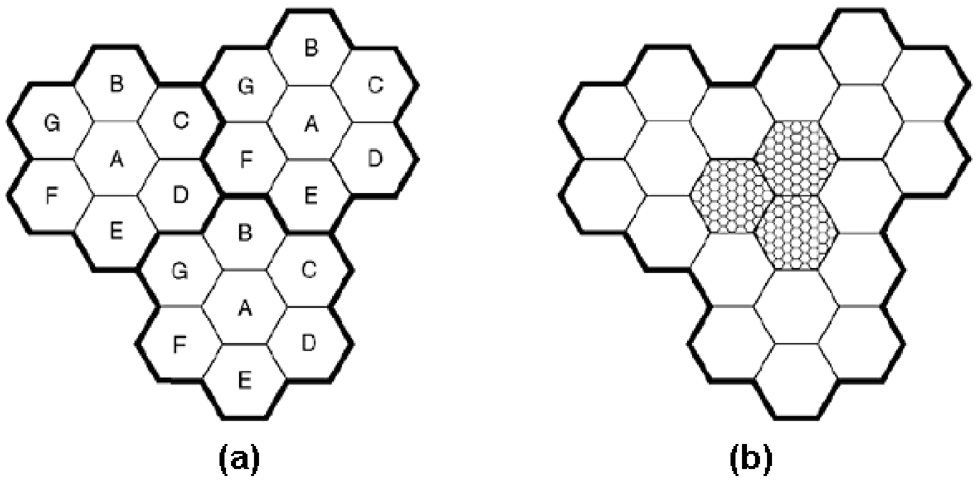

Figure 1.2

(a) Frequencies are not reused in adjacent cells.

(b) To add more users, smaller cells can be used (The cells are normally roughly circular).

Cells are grouped in units of seven cells. Each latter indicates a group of frequencies. Notice that for each frequency set, there is a buffer about two cells wide where that frequency is not reused, providing for good separation and low interference. The base station consists of a computer and transmitter/receiver connected to an antenna. In a small system, all the base stations are connected to a single device called an MSC (Mobile Switching Center). In a larger one, several MSC may be needed, all of which are connected to a second-level MSC, and so on.

The MSCs communicate with the base stations, and the Public Switching Telephone Network (PSTN) using a packet switching network. At any instant, each mobile telephone is logically in one specific cell and under control of that cell's base station. When a mobile telephone physically leaves a cell, its base station notices the telephone's signal finding away and asks the surrounding base stations how much power they are getting from it. The base station then transfers ownership to the cell getting the strongest signal, that is, the cell where the telephone is now located. The telephone is then informed of its new boss, and if a cell is in progress, it will be asked to switch to a new channel (because the old is not reused in any of adjacent cells). This process, called handover takes about 300 msec.

Handover can be done in two ways, in a soft handover; the new base station acquires the telephone before the previous one signs off, in this way, there no loss of continuity. The downside here is that the telephone needs to be able to tune to two frequencies at the same time (the old one and the new one). Neither first nor second-generation devices can do this (available only with CDMA).

In a hard handover, the old base station drops the telephone before the new one acquires it. If the new one is unable to acquire it (e.g., because there is no available frequency), the cell is disconnected abruptly. Users tend to notice this, but it is inevitable occasionally with the current design [8], [9], [10].

1.4.1.2 Channels

The AMPS system uses 832 full-duplex channels, each consisting of a pair of simplex channels. There are 832 simplex transmission channels from 824 to 849 MHz and 832 simplex receive channels from 869 to 894 MHz. Each of these simplex channels is 30 KHz wide. Thus, AMPS uses FDM to separate the channels.

In the 800-MHz band, radio waves are about 40 cm long and travel in straight lines. They are absorbed by trees and plants and bounce off the ground and buildings. It is possible that a signal sent by a mobile telephone will reach the base station by the direct path, but also slightly later after bouncing off the ground or a building. This may lead to an echo or signal distortion (multi-path fading).

The 832 channels are divided into four categories:

1. **Control -** (base to mobile) to manage the system. (21 channels).
2. **Paging -** (base to mobile) to alert mobile users to call for them.
3. **Access -** (bidirectional) for call setup and assignment.
4. **Data -** (bidirectional) for voice, fax, or data.

1.4.1.3 Call Management

Each mobile telephone in AMPS has a 32-bit serial number and a 10-digit telephone number in its PROM. The telephone number is represented as a 3-digit area code in 10 bits, and a 7-digit subscriber number in 24 bits. When a phone is switched on, it scans a preprogrammed list of 21 control channels to find the most powerful signal. The phone then broadcasts its 32-bit serial number and 34-bit telephone number. Like all the control information in AMPS, this packet is sent in digital for, multiple times, and with an error-correcting code. When base station hears the announcement, it tells the MSC, which records the existence of its new costumer and also informs the costumer’s home MSC of his current location. During normal operation, the mobile telephone registers about once every 15 minutes.

To make a call, a mobile user switches on the phone, enters the number to be called on the keypad, and hits the SEND button. The phone then transmits the number to be called and its own identity on the access channel. If a collision occurs there, it tries again later. When the base station gets the request, it informs the MSC. If the caller is a costumer of the MSC's company (or one of its partners), the MSC looks for an idle channel for the call. If one is found, the channel number is sent back on the control channel. The mobile phone then automatically switches to the selected voice channel and waits until the called party picks up the phone.

1.4.2 Second Generation Mobile Network (Digital Voice)

The second generation was digital and four systems are in use now; D-AMPS, GSM, CDMA, and PDC. Below discuss the first two. PDC is used only in Japan (is basically D-AMPS modified) and for backward compatibility with the first-generation Japanese analog system [7], [8], [9], [10].

1.4.2.1 D-AMPS- The Digital Advanced Mobile Phone System

D-AMPS is fully digital. It is described in International Standard IS-54 and its successor IS-136. D-AMPS designed to co-exist with AMPS so that both first and second-generation mobile phones could operate simultaneously in the same cell. In particular, DAMPS uses the same 30 kHz wide channels as AMPS and at the same frequencies so that one channel can be analog and the adjacent ones can be digital. The cell's MSC determines which channels are analog and which are digital. When D-AMPS was introduced, a new frequency band was made; for upstream channel were in the 1850-1910 MHz range, and the corresponding downstream channels was in the 1930-1990 MHz range, as in AMPS. In this band the waves are 16 cm long, so a standard ¼-wave antenna is only 4 cm long, leading to smaller phones. Many D-AMPS phones can use both the 850 MHz and 1900 MHz bands to get a wider range of available channels.

On a D-AMPS mobile phone, the voice signal picked up by the microphone is digitized and compressed. Compression is done from the standard 56-kbps PCM encoding to 8 kbps, or less. The compression is done in the telephone. With mobile phone there is a huge gain from doing digitizing and compression in the handset, so, in D-AMPS, three users can share a single frequency pair using time division multiplexing. Each frequency pair supports 25 frames/sec of 40 msec each. Each frame is divided into six time slots of 6.67 msec each, see Figure 1.3.

During slot 1 Figure 1.3 (a), user 1 may transmit to the base station and user 3 is receiving from the base station. With 50 slot/sec, the bandwidth available for compressed speech is just less than 8 kbps, 1/7 of the standard PCM bandwidth. Using better compression algorithms, it is possible to get the speech down to 4 kbps, in which case six users can be stuffed into a frame. Of course, the quality of speech at 4 kbps is not comparable to what can be achieved at 56 kbps.

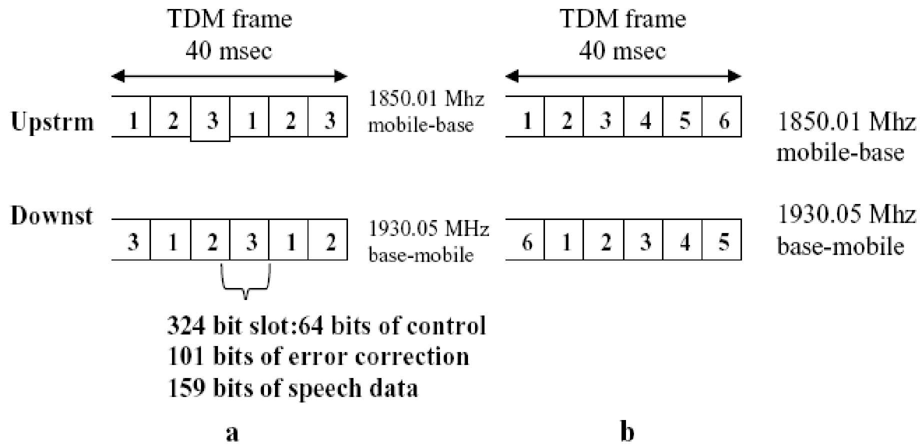

Figure 1.3

(a) A D-AMPS channel with 3 users (b) A D-AMPS channel with 6 users

The control structure of D-AMPS is complicated. Groups of 16 frames form a super frame, with certain control information present in each super frame a limited number of times. Six main control channels are used: system configuration, real-time and non real-time control, paging, access response, and short messages. But conceptually, it works like AMPS. When a mobile is switched on, it makes contact with the base station to announce itself and then listens on a control channel for incoming calls. Having picked up a new mobile, the MSC informs the user's home base where he is, so calls can be routed correctly. One difference between AMPS and D-AMPS is how handover is handled. In AMPS, the MSC manages it completely without help from mobile devices.

As can be seen from Figure 1.3, in D-AMPS, 1/3 of the time a mobile is neither sending nor receiving. It uses these idle slots to measure the line quality. As in AMPS, it still takes about 300 msec to do the handover. This technique is called MAHO (Mobile Assisted Handover).

1.4.2.2 GSM-The Global System for Mobile Communications

D-AMPS is widely used in the U.S. and in Japan (in modified form). Virtually everywhere else in the world, a system called GSM (Global System for Mobile Communications) is used, and it is even starting to be used in the U.S. on a limited scale.

To a first approximation, GSM is similar to D-AMPS. Both are cellular systems. In both systems, FDM is used, with each mobile transmitting on one frequency and receiving on a higher frequency. In both systems, a single frequency pair is split by TDM into time slots shared by multiple mobiles. However, the GSM channels are much wider than the AMPS channels (200 kHz versus 30 kHz). GSM has much higher data rate per user than D-AMPS. GSM frequency band (200 kHz) is shown in Figure 1.4 [7].

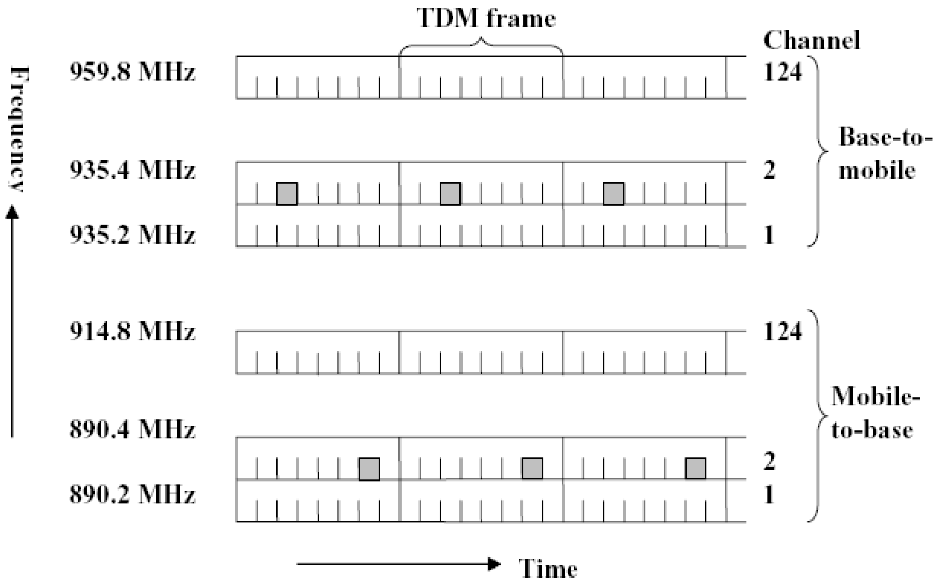

Figure 1.4: GSM uses 124 frequency simplex channels, each of which uses an eight-Slot TDM system.

A GSM system has 124 pairs of simplex channels; each simplex channel is 200 kHz wide and supports eight separate connections on it, using TDM. Each currently active station is assigned one time slot on one channel pair. Theoretically, 192 channels can be supported in each cell, but many of them are not available, to avoid frequency conflict with neighboring cells. The eight (in the figure is absent one TDM frame with one time slot) shaded time slot all belong to the same connection, four of them in each direction.

Transmitting and receiving does not happen in the same time slot because the GSM radios cannot transmit and receive at the same time and it takes time to switch from one to the other. If the mobile station assigned to 890.4/935.4 MHz and time slot 2 wanted to transmit to the base station, it would use the lower four shaded slots (and the ones following them in time), putting some data in each slot until all the data had been sent. The TDM slots shown in Figure 1.4 are part of a complex framing hierarchy. Each TDM slot has specific structure, and groups of TDM slots form multi frames, also with a specific structure. A simplified version of this hierarchy is shown in Figure 1.5.

Here, each TDM slot consists of a 148-bit data frame that occupies the channel for 577 microseconds (including a 30- microsecond. guard time after each slot). Each data frame starts and ends with three 0 bits, for frame description purposes. It also contains two 57-bit information fields, each one having a control bit that indicates whether the following information field is for voice or data. Between the information fields is a 26-bit Sync (training) field that is used by the receiver to synchronize to the sender’s frame boundaries.

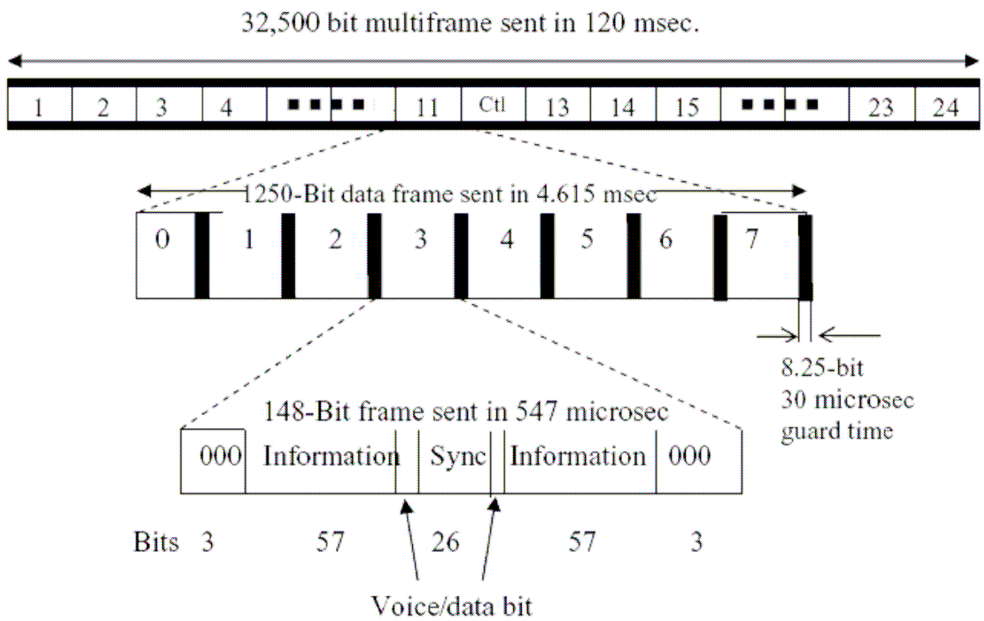

Figure 1.5: A portion of the GSM framing structure

A data frame is transmitted in 547 microsec, but a transmitter is only allowed to send one data frame every 4.615 msec, since it is sharing the channel with seven other stations. The gross rate of each channel is 270,833 bps, divided among eight users. This gives 33.854 kbps gross, more than double D-AMPS, 324 bits, 50 times per second for 16.2 kbps. However, as with AMPS, the overhead eats up a large fraction of the bandwidth, ultimately leaving 24.7 kbps worth of payload per user before error correction. After error correction, 13 kbps is left for speech, giving substantially better voice quality than D-AMPS (at the cost of using correspondingly more bandwidth).

As it is shown in Figure 1.5, eight data frames make up a TDM frame and 26 TDM frames make up a 120-msec multiframe, of the 26 TDM frames in a multiframe, slot 12 is used for control and slot 25 is reserved for future use, so only 24 are available for user traffic.

1.4.3 Third Generation Mobile Network (Digital Voice and Data)

In 1992, ITU issued a blueprint for getting there called IMT-2000, where IMT stood for International Mobile Telecommunications. The number 2000 stood for three things:

1. The year it was supposed to go into service.
2. The frequency it was supposed to operate at (in MHz).
3. The bandwidth the service should have (in kHz).

It did not make it on any of three counts. Nothing was implemented by 2000. ITU recommended that all governments reserve spectrum at 2 GHz so devices could roam seamlessly from country to country. Finally, it was recognized that 2 Mbps is not currently feasible for users who are too mobile (due to the difficulty of performing handover quickly enough). More realistic is 2 Mbps for stationary indoor users, 384 kbps for people walking, and 144 kbps for connections in cars. Nevertheless, the whole area of 3G as it is called is one great activity. The 3G may be a bit less than originally hoped for and a bit late, but it is happening [7], [8], [9], [10].

The basic services that the IMT-2000 network is supposed to provide to its users are:

1. High-quality voice transmission.
2. Messaging (replacing e-mail, fax, chat, etc.).
3. Multimedia (playing music, viewing videos, films, televisions, etc.).
4. Internet access (Web surfing, including pages with audio and video).

All these services are supposed to be available worldwide (with automatic connection via a satellite when no terrestrial network can be located). Several proposals were made; they came down to two main ones.

- The first one, W-CDMA (Wideband CDMA), was proposed by Ericsson. This system uses direct sequence spread spectrum. It runs in a 5 MHz bandwidth and has been designed to interwork with GSM networks although it is not backward compatible with GSM. It does, however, have the property that a cellular can leave a W-CDMA cell and enter a GSM cell without losing the call. This system called UMTS (Universal Mobile Telecommunication System).
- The other contender was CDMA2000, proposed by Qualcomm. It, too, is a direct sequence spread spectrum design, basically an extension of IS-95 and backward compatible with it. It also uses a 5-MHz bandwidth, but it has not been designed to interworking with GSM and cannot hand off calls to a GSM cell (or D-AMPS). Other technical differences with W-CDMA include a different chip rate, different frame time, different spectrum used, and different way to do time synchronization.

While waiting for the fighting over 3G to stop, some operators are gingerly taking a cautious small step in the direction of 3G by going to what is sometimes called 2.5G, although 2.1G might be more accurate. One such system is EDGE (Enhanced Data rates for GSM Evolution), which is just GSM with more bits per baud. The trouble is, more bits per baud also means more errors per baud, so EDGE has nine different schemes for modulation and error correction, differing on how much of the bandwidth is devoted to fixing the errors introduced by the higher speed.

Another 2.5G scheme is GPRS (General Packet Radio Service), which is an overlay packet network on top of D-AMPS or GSM. It allows mobile stations to send and receive IP packets in a cell running a voice system. When GPRS is in operation, some time slots on some frequencies are reserved for packet traffic. The base station can dynamically manage the number and location of the time slots, depending on the ratio of voice to data traffic in the cell.

The available time slots are divided into several logical channels, used for different purposes. The base station determines which logical channels are mapped onto which time slots. One logical channel is for downloading packets from the base station to some mobile station, with each packet indicating who it is destined for. To send an IP packet, a mobile station requests one or more time slots by sending a request to the base station. If the request arrives without damage, the base station announces the frequency and time slots allocated to the mobile for sending the packet. Once the packet has arrived at the base station, it is transferred to the Internet by a wired connection. Since GPRS is just an overlay over the existing voice system, it is at best a stop-gap measure until 3G arrives.

3G (third-generation) mobile systems are defined by International Telecommunications Union (ITU) specification IMT-2000 (International Mobile Telecommunications-2000), a radio and network access specification. 3G are the successor of 2G—the existing and hugely deployed digital mobile system. 2G are the successor of 1G, the original analogue mobile system. GSM is the most predominant choice for 2G deployments. Though voice remains the primary method of mobile communication, a new generation of wireless technologies is now offering higher speed data and multimedia capabilities.

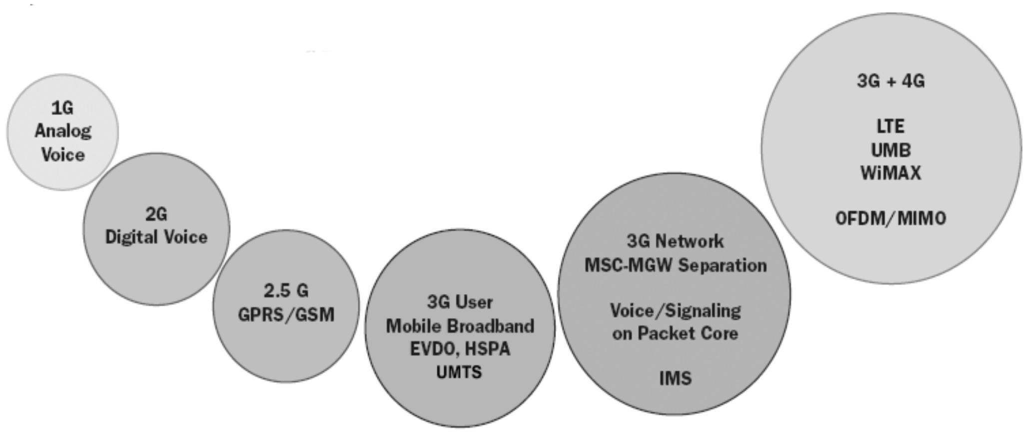

Figure 1.6: Cellular Network Evolution: 1G to 4G

If the request arrives without damage, the base station announces the frequency and time slots allocated to the mobile for sending packet. Once the packet has arrived at the base station, it is transferred to the Internet by a wired connection. Since GPRS is just an overlay over the existing voice system, it is at best a stop-gap measure until 3G arrives. Even though 3G are not deployed fully yet, some researchers regard 3G as a done deal and thus not interesting any more.

1.5 Footprint towards Next Generation Networking

The mobile industry has witnessed explosive growth in number of subscribers, particularly over the past few years. However, while usage measured in terms of the number of wireless minutes is increasing, the price per minute for these services is falling.

This means that Average Revenue Per User (APRU) is shrinking. Running a profitable business with stagnant or even declining ARPU is one of the fundamental challenges mobile carriers are facing today. The industry is addressing this challenge in two ways:

- By adding new services or new user experiences for which mobile subscribers are willing to pay.
- By reducing operating expenses (OPEX). At the top of the list is the wireline infrastructure that mobile operators have to maintain regardless of whether they own or lease lines.

Today, voice still accounts for the majority of overall cellular traffic, with wireless data exceeding more than 10 percent of mobile operator ARPU. Mobile operator ARPU is under pressure due to price and technology competition from both wireline (for example, voice over IP) and emerging services (for example, voice over Wi-Fi). Although mobile operator ARPU for voice services is declining, the ARPU for data revenues is growing at a healthy rate.

Standards bodies such as 3GPP (for GSM networks) and 3GPP2 (for CMDA networks) are actively involved in driving the development of a next-generation wireless system. The high level objective is to create high-speed broadband and IP-based mobile systems featuring network-to network interconnection, feature/service transparency, global roaming, and seamless services independent of location.

Telecommunications industry is at a crossroads. It was till yesteryears that many technologies came into play in telecom industry. Some of these evolved with time and some started new. All these efforts to make telecommunication more secured, effective, consumer friendly, and cost effective. Recently a new trend emerged to provide a complete solution for convergence, interoperability, and security. IMS redefined the inter-working of different networks and provides a user friendly, cost effective solution for the consumers. It made a big impact as it also addresses the problems faced by service providers and mobile operators through standard specifications to reduce operational cost, roaming problems, and many others. IMS is a big vision of the future of telecommunications it hangs on.

At present, cellular telephone networks provide services to over one billion users worldwide as on today. These services include, of course, telephone calls, but are not limited to them. Modern cellular networks provide messaging services ranging from simple text messages (e.g., SMS, Short Messaging Service) to fancy multimedia messages that include video, audio, and text (e.g., MMS, Multimedia Messaging Service). Cellular users are able to surf the Internet and read email using data connections, and some operators even offer location services, which notify users when a friend or colleague is nearby.

Still, cellular networks did not become so attractive to users only for the services they offered. Their main strength is that users have coverage virtually everywhere. Within a country, users can use their terminals not only in cities, but also in the countryside. In addition, there exist international roaming agreements between operators that allow users to access cellular services when they are in roaming.

Reduction in Mobile Equipment size also helped the spread of cellular networks. Old Mobile terminals gave way to modern small terminals that work several days without having their batteries recharged. This allows people to carry their terminals everywhere with little difficulty.

NGN is a concept that has been introduced to take into account the new situation and changes in the telecommunications fields. This new situation is characterized by a number of aspects: the deregulation of markets, the new demand from users for innovative services to meet there needs, and the explosion of digital traffic (increase of Internet usage). The introduction of NGN comprises economic and technical aspects. Economically, it allows increasing productivity by creating new usage [11] based on user preferences and related to voice and data services (e.g., voice over IP, instant messaging, presence, streaming, and push to talk). It also permits reducing costs for infrastructure maintenance, with only one type of transport network instead of specific ones for each access network. Technically, NGN makes the network architecture flexible in order to define and introduce new services easily.

The cornerstone of the service architecture for next-generation networks is the IMS architecture, standardized by 3GPP. The IMS offers telecom operators the possibility to build an open IP-based service infrastructure that will enable easy deployment of new, rich multimedia communication services mixing telecom and data services.

The conception of IMS services is a key challenge for the telecom market. IMS services are fundamentally tailored to user preferences, rely seamlessly on multiple access networks, and bundle multiple service features (e.g., voice/video connectivity, community tools, presence, conferencing, gaming, and TV broadcasting). The architecture and technical aspects of the IMS architecture are well addressed by the standardization bodies. However, a clear model of what an IMS service not proposed by these bodies. The objective of this chapter is to detail the concepts behind IMS services and to propose a way to link IMS service, service building blocks, and technical functions.

1.5.1 From IN to NGN

The concept of intelligent networks (INs) developed in the 1980s was a precursor of the NGN. The principle of INs is to separate clearly the switching functions from the service data and logic located in an external entity: the Service Control Point (SCP). A new functional entity is added to the TDM (Time Division Multiplexing) switch, the Service Switching Point (SSP), which interfaces between the service logic and the switch itself. An interface based on the Intelligent Network Application Part (INAP) protocol family is introduced between the SSP and the SCP. The services are no longer developed in the TDM switch—as with the concept of global system for mobile communications (GSM) and integrated services digital network (ISDN) supplementary services—but rather are implemented in the SCP. The INAP and associated procedures allow the SCP to control and monitor the switch.

The intelligent network introduced the concept of a Service Independent Building Block (SIB) for reusable service functions. A service could thus be thought of as a composition of various SIBs. But this goal was not fully achieved because of a lack of independence with INAP protocol, a lack of software reusability, and a lack of openness by manufacturers and operators. As a consequence, INs deployed today relies on a monolithic architecture and service platforms do not offer flexible services. In addition, as the service logic is executed in external entities, triggering multiple services for one call requires having service interaction management mechanisms.

This issue, known as feature interaction, is one of the most complex problems encountered in IN and considerable work has been done on it. However, this work cannot be directly applied to the NGN because of the service and architectural differences between IN and NGN. The promise of the NGN, as defined in the late 1990s, was to offset these shortcomings by moving from a vertical approach (where access, control, and services are closely tied) to a horizontal approach (where each layer provides reusable elements to other layers).

Specification work is ongoing at the International Telecommunication Union (ITU)-T to formalize the separation (e.g., through standard protocols or application programming interfaces [APIs]) between the transport stratum that is composed of transfer functions from various access networks (UMTS terrestrial radio access network [UTRAN], wireless local area network [WLAN], xDSL) and from the core networks, control functions for these transfer functions (e.g., network attachment control or resource and admission control), the transport user profiles (e.g., to store the data linked to network attachment), and the media handling functions (e.g., for playing announcements or for transcoding); and the service stratum composed of access-independent service control functions (e.g., session establishment control or service triggering control), application functions, and service user profiles. Application functions should be independent from the service control functions and should offer flexibility (e.g., by using open software mechanisms) to answer user needs. [12]

1.6 World before IMS

Till early 2000, operators were facing lot of difficulties in interconnecting networks working on different technologies. With these consumers, when outside the home network was made to pay a big amount towards roaming charges, these overhead charges increased as the number of mobile users increased. Though technology evolved to a great extent, service providers were not ready to go for “rip and replace” option and invest a huge amount to have a 3G system in place.

The GSM system which started as 2G technology evolved into GPRS (also known as 2.5G), then to a 3G called UMTS (Universal Mobile Telecommunications System) Almost at the same time a new technology came into existence called CDMA (Code Division Multiple Access). This system developed with a greater speed in United States and evolved to 3G systems called CDMA2000 1xEV-DO and then to CDMA2000 1xEVDV. In IP domain people were moving from H323 to all SIP and RTP based systems emphasizing more on security, bandwidth utilization, and 3G features. IMS came as a Next Generation Network NGN to address these issues with Service oriented architecture.

1.6.1 Requirement of Emerging Technology

Need to further clarify what does it mean by merging the Internet and the cellular worlds and what the real advantages of doing so are. To do that, need to introduce the different domains in 3G networks, namely the circuit-switched domain and the packet switched domain.

The circuit-switched domain is an evolution of the technology used in Second Generation (2G) networks. The circuits in this domain are optimized to transport voice and video, although they can also be used to transport instant messages. Although circuit-switched technology has been in use since the birth of the telephone, the current trend is to substitute it with more efficient packet-switched technology. Cellular networks follow this trend and, as said earlier, 3G networks have a packet-switched domain.

The packet-switched domain provides IP access to the Internet. While 2G terminals can act as a modem to transmit IP packets over a circuit, 3G terminals use native packet switched technology to perform data communications. This way, data transmissions are much faster and the available bandwidth for Internet access increases dramatically. Users can surf the web, read email, download videos, and do virtually everything they can do over any other broadband Internet connection, such as ISDN (Integrated Services Digital Line) or DSL (Digital Subscriber Line). This means that any given user can install a VoIP client in their 3G terminals and establish VoIP calls over the packet-switched domain. Such a user can take advantage of all the services that service providers on the Internet offer, such as voice mail or conferencing services.

The needs for New Technology for Mobile communication can be categorized in three way [13], [14]:

- QoS (Quality of Service),
- Charging,
- Integration of different services.

The main issue with the packet-switched domain to provide real-time multimedia services is that it provides a best-effort service without QoS; that is, the network offers no guarantees about the amount of bandwidth a user gets for a particular connection or about the delay the packets experience. Consequently, the quality of a VoIP conversation can vary dramatically throughout its duration. At a certain point the voice of the person at the other end of the phone may sound perfectly clear and instants later it can become impossible to understand. Trying to maintain a conversation (or a videoconference) with poor QoS can soon become a nightmare.

1.7 Introduction to NGN

The Next Generation Network (NGN) concept defines telecommunications network architectures and technologies. It describes networks that cover conventional PSTN (Public Switched Telephone Network) type of data and voice communications as well as new types of service such as video.

All information is carried in packet switched form, as is done in the Internet. Packets are labeled according to their type (data, voice, video, etc) and forwarded in the network based on their Quality of Service (QoS) and security parameters. The NGN makes a clear separation between the transport and services, which is advertised to allow smooth introduction of new services. When a provider wants to launch a new service, the service is defined directly at the service layer without considering the transport layer, i.e. services are independent of the transport technology.

1.7.1 IMS Introduction

IMS is a global, access-independent and standard-based IP connectivity and service control architecture that enables various types of multimedia services to end-users using common Internet-based protocols. IMS architecture is at the heart of the convergence of voice, data, and fixed and mobile networks and is based on a wide range of IETF protocols, such as SIP. IMS combines and enhances these protocols to allow real-time services in addition to 3GPP mobile Packet-Switched (PS) domain and the wireline NGN.

After having released their first specification in 1999 (3GPP R99), the 3GPP started to specifying Release 2000, which included the All-IP systems, that was later renamed the IMS. After realizing that the development of IMS could not be completed during the year 2000, the agreement was reached to split Release 2000 into Release 4 and Release 5 specifications. After Release 4 has been frozen and completed, IMS was introduced by the 3GPP within Release 5 in the year 2002.

The IMS specified in Release 5 was only based on the 3G mobile systems, and only compatible to IP version 6 (IPv6). The 3GPP continued their IMS specifications and enhancements in Releases 6 and 7 in year 2007 of their specifications. Release 6 IMS [19] included the interworking with the CS networks and other IP based networks. It also introduced more service provisioning, which included Presence service; Push to-talk-over Cellular (PoC) and Instant Messaging (IM) servers. IMS release 7 specifications that was released in 2006 included further IMS enhancements.

SIP allows applications to remain agnostic of the access network, which matches the network access requirements for IMS. Initial concepts for IMS emerged with the UMTS 3G specifications in 1998. The first specification of IMS was published in March of 2003 by 3GPP in UMTS Release 5. UMTS Release 5 provided general description of IMS, SIP and end-to-end Quality of Service as part of an “All IP” network. IMS continues to evolve with each UMTS release since 2003. New functions and changes to IMS are introduced through 3GPP approved change requests (CRs) with each release. 3GPP release 6 and 7 added interworking with wireless local area networks (WLAN) and support for fixed networks, by working together with TISPAN (Telecom & Internet converged Services & Protocols for Advanced Networks).

The IP Multimedia Subsystem (IMS) is a standardized Next Generation Networking (NGN) architecture for telecom operators that want to provide mobile and fixed multimedia services. It uses a Voice-over-IP (VoIP) implementation based on a 3GPP-standardized implementation of SIP, and runs over the standard Internet Protocol (IP). Existing phone systems (both packet-switched and circuit-switched) are supported.

So, one of the reasons for creating the IMS was to provide the QoS required for enjoying, rather than suffering, real-time multimedia sessions. The IMS takes care of synchronizing session establishment with QoS provision so that users have a predictable experience.

Another reason for creating the IMS was to be able to charge multimedia sessions appropriately. A user involved in a videoconference over the packet-switched domain usually transfers a large amount of information (which consists mainly of encoded audio and video). Depending on the 3G operators the transfer of such an amount of data may generate large expenses to the user, since operators typically charge based on the number of bytes transferred. The user's operator cannot follow a different business model to charge the user because the operator is not aware of the contents of those bytes: they could belong to a VoIP session, to an instant message, to a web page, or to an email.

1.7.2 IMS Standardization bodies

There are several internationally recognized telecommunication and Internet standardization bodies responsible for the standardization of the IMS, SIP and Java technologies. This section cites major IMS, SIP and Java standardization bodies.

1.7.2.1 The 3rd Generation Partnership Project (3GPP)

According to the partnership agreement signed in December 1998, the 3GPP was formed in 1998 by the standardization bodies from Europe, Japan, USA and China to establish a 3rd generation (3G) mobile system based on evolved GSM core networks and the radio access technologies. The main purpose of 3GPP is to prepare, approve and maintain globally applicable Technical Specifications (TS) and Technical Reports (TR) for 3G mobile systems. IMS was introduced in Release 5 [19] as part of 3GPP UMTS specification, and was mainly dedicated for 3G mobile systems. Release 6 and beyond of 3GPP specification added wireline support to IMS architecture.

1.7.2.2 European Telecommunications Standard Institute (ETSI)

ETSI has established the Telecoms and Internet converged for Services and Protocols for Advanced Network (TISPAN) project as their core competence centre for the Next Generation Networks (NGN). TISPAN facilitates the smooth migration of fixed (wireline) networks from Circuit-Switched (CS) networks to Packet-Switched (PS) networks with an architecture that can serve in both to create the NGN. TISPAN has adopted the IMS architecture given in the 3GPP Release 6 but has added wireline access to the IMS [15]. Essentially, TISPAN adopts the 3GPP IMS architecture standard for SIP-based applications, but has added further functional blocks and subsystems to handle non-SIP applications and other requirements of fixed networks not addressed by IMS.

1.7.2.3 International Telecommunications Union (ITU)

ITU-T created an NGN Focus Group (NGN-FG) in May 2004 to work on the specifications of NGN for fixed line access based on 3GPP IMS. The process of fitting IMS into ITU's NGN is performed by the ETSI TISPAN group, which is in turn responsible for all aspects of standardization for present and future converged networks [16].

1.7.2.4 The Internet Engineering Task Force (IETF)

The IETF is an open forum consisting of individuals working with vendors, operators, researchers and other interested individuals who make technical and other contributions to the engineering and evolution of the Internet and its technologies [17]. There are two documents used within the IETF, namely the Internet-Drafts (I-Ds) and the Request for Comments (RFC). A standard that begins life as an I-D, progresses to an RFC once there is consensus and there are working implementations of the protocol. When changes are made in a protocol or new versions come out, a new RFC document with a new number is issued, which renders the old RFC obsolete.

The 3GPP have established collaboration with the IETF to make sure that the protocols developed for IMS meet their requirements. This collaboration has been documented in RFC 3113. The IETF is responsible for the standardization of SIP, RFC 3261 being the core SIP specification. Several RFCs were developed as the extension to the core RFC 3261 in order to meet 3GPP IMS requirements. The selection of SIP as the core IMS signalling protocol makes the IMS network to be regarded as an advanced SIP network.

1.7.2.5 Java Community Process (JCP)

Specifications for Java platforms are developed under the aegis of the JCP. A specification starts off as a Java Specification Request (JSR). An expert group consisting of representatives from interested companies is formed to create the specification. The JSR then passes through various stages in the JCP before it is finalized. Every JSR is assigned a number. The JCP is also involved in the standardization of Java APIs to facilitate the easy and fast deployment of IP services using SIP. The following are JSRs defined by JCP for SIP:

- **JAIN SIP:** JSR 32 determines the Java Advanced Intelligent Network (JAIN) SIP specification. Sun Microsystems is the specification lead in conjunction with other expert companies and individuals. JAIN SIP can be implemented on Personal Digital Assistance (PDA), SIP phones and desktops.

- **SIP Servlet:** JSR 116 defines the SIP Servlet specification. Dynamicsoft is the specification lead in conjunction with other expert companies. SIP Servlet is implemented in web tier enterprise application servers, which has the benefit of converged SIP and HTTP applications.
- **SIP for J2ME:** JSR 180 determines the SIP for J2ME specification - SIP interface for small platforms. Nokia Corporation is the specification lead in conjunction with other expert companies and individuals. SIP's acceptance as the protocol of choice by the IMS

1.7.3 IMS vision

IMS is at the center of the 3G initiatives driven by 3GPP and 3GPP2 who are participants in International Mobile Telecommunications-2000 (IMT-2000) the global standard for third generation wireless communications, defined by a set of interdependent ITU Recommendations.

The 3GPP was formed as a collaboration of many organizations (includes Association of Radio Industries and Businesses, China Communications. Standards Association, European Telecommunications Standards Institute, Alliance for Telecommunications Industry Solutions, Telecommunications Technology Association of South Korea and Telecommunication Technology Committee of Japan) in 1998 to develop the technical specifications for a 3G network evolving from a GSM network. A similar charter, 3GPP2 (includes Association of Radio Industries and Businesses, Telecommunication Technology Committee of Japan, China Communications. Standards Association, Telecommunications Technology Association of South Korea and Telecommunications Industry Association), was formed to evolve the North American and Asian networks from traditional CDMA2000 networks into a 3G platform.

The main vision of IMS is to bring the Fixed Mobile Convergence (FMC) to reality by enabling the provision of multimedia services to both fixed and mobile subscribers. This vision has been achieved by bridging the gap that exists between the traditional telecommunications provided by both cellular and Public Switched Telecommunication Network (PSTN) operators and the Internet. IMS provides the key functionalities required to enable new IP services via mobile networks, taking into account the complexity of managing mobility, multimedia, constraints of the underlying network and the multitude of emerging applications. With IMS, competition between fixed and mobile users will exist because all the service providers and operators will be targeting a single customer to offer similar services.

Both 3GPP and 3GPP2 conceptualized their respective IMS architectures to support packet switched communication, in order to merge the Internet and the cellular worlds. The IMS core network has a common IP based transport and signaling, which can be accessed by different networks. IMS common signaling is based on SIP

The vision of IMS as the common platform for development and delivery of diverse multimedia services for a true mobile Internet is based on the set of requirements set forth in the 3GPP IMS requirements captured in 3GPP TS 22.228 [19].

1.7.4 IMS Current Scenario

The IP Multimedia Subsystem standard defines a generic architecture for offering Voice over IP and multimedia services. It is an international, recognized standard, first specified by the Third Generation Partnership Project (3GPP/3GPP2) and now being embraced by other standards bodies including ETSI/TISPAN. The standard supports multiple access types – including GSM, WCDMA, CDMA2000, Wireline broadband access and WLAN. For users, IMS-based services enable person-to-person and person-to-content communications in a variety of modes – including voice, text, pictures and video, or any combination of these – in a highly personalized and controlled way.

For operators, IMS takes the concept of layered architecture one-step further by defining a horizontal architecture, where service enablers and common functions can be reused for multiple applications. The horizontal architecture in IMS also specifies interoperability and roaming, and provides bearer control, charging and security. What is more, it is well integrated with existing voice and data networks, while adopting many of the key benefits of the IT domain.

This makes IMS a key enabler for fixed-mobile convergence (FMC). For these reasons, IMS will become preferred solution for fixed and mobile operators' multimedia business. IMS enables services to be delivered in a standardized, well-structured way that truly makes the most of layered architecture. At the same time, it provides a future-proof architecture that simplifies and speeds up the service creation and provisioning process, while enabling legacy interworking. The horizontal architecture of IMS enables operators to move away from vertical 'stovepipe' implementations of new services – eliminating the costly and complex traditional network structure of overlapping functionality for charging, presence, group and list management, routing and provisioning. For fixed and mobile operators there are benefits of introducing the IMS architecture today. On longer term, IMS enables a secure migration path to an all-IP architecture that will meet end-user demands for new enriched services.

1.8 Research Methodology

A qualitative research methodology with action research will be used for conducting this research work. Action research is appropriate and has been chosen not to incline towards the results, as this is a step-by-step implementation studied project, also studied and analyzed the real reasons behind the results [18].

To float application on any existing working network is an approach of simulation of research work. But considering own effort for this research and aim to perform economical and inclusive research, whole research starting from scratch to the highest possible level, this research initiates from configuration and implementation of each entity of IP Multimedia Subsystem as well as related components created and for capturing results and evaluate performance mainly freeware and open source software chosen.

This research also involves in improving an existing situation with considering future trends in service provision in Communication network. Extensive literature survey and analytical thinking are involved in this research project.

At first, an understanding of problem is developed through an intensive qualitative literature study. An in-depth literature study of the 3GPP IMS by reading the Technical Specifications, books and published work, related to the IMS was conducted. By applying analytical research, the literature collected is thoroughly analyzed in Chapter 3.

First phase of create ground for IMS, in this research, Open IMS core test bed setup established, with considering its SIP initialization as well as evaluate each logs for session through different stage of work over IMS.

With second phase over IMS Test bed, a light-size IMS client developed using J2ME and supportive tools with basic concept to reduce size of IMS client and make maximum number of application float over it. Java Specification Request (JSR) number 180 (JSR 180) was identified and studied as the core SIP API to be used for the development of the IMS Client. Sun Wireless Toolkit was used as the Java 2 Micro-Edition (J2ME) development platform for the client. EclipseME Integrated Development Environment (IDE) was used to edit the Java code. Moreover in this research all existing IMS clients are studied but the low-size IMS client designed to float over Open IMS Core Test Bed.

The next phase of research approached Qualitative Research methodology by implementation of Instant Messaging and Presence service over Open IMS Core Test Bed, with continuation of this phase of Qualitative Research observation of Presence Server Load Balancing and Traffic Analysis made over Open IMS Core Test Bed. Considering results a Load Management study covering for future Mobile network management.

In fourth Phase, the architecture evaluation in chapter 7 is also a type of analytical research in which concept provided by action research is analyzed. Several state-of-the-arts proposed applying action research leading to propose simple and flexible security architecture to IMS SIP scrutinize solutions. In later stages quantitative analysis is involved in architecture evaluation like response time calculation.

During literature review, that was analyzed for new and innovative application creation and float over IMS Network, considering this during the Fifth phase in this research a new and light weight Mobile based Blogging application is created and implemented over Open IMS Core Test Bed. This Qualitative approach demonstrates future approach to float new and strategically applications over IMS network and prototyping the same. With MoBlog application the concept of new service design, deployment and modeling demonstrates Qualitative approach over IMS Network.

Figure 1.7 (in next page) shows brief overview and flow of research methodology for this research work.

1.8.1 Audience

Research work presented in this thesis is relevant to those people who have knowledge of Mobile Communication Technology, IP Multimedia Subsystem, SIP, Next Generation Networking. This research reveal to be guideline for novice in the area of NGN research, as this research work consisting all basic to advance stage of IMS related work ranging from Network setup Test bed formation, SIP initiation and Service modeling.

For experts in the area of Next Generation Network, advance part of research presented in this Thesis able to make their research relevance to service discusses in this work. This service and results seems to be guideline for new and further enhancement of research in the IMS and convergence world.

1.9 Benefits of the Study

This research work discuss persistent work over IMS for more than 2.5 years, parallel to this research a lot more work in other part of world also done. Before initiation of this research a deep root study of existing work made and then from that all, this research carried to proved unique and contributed a new results, discussion and service modeling techniques to the World of Next Generation Networks.

- This research work presents IMS implementation strategy with SIP through Open IMS Core and forming Open IMS Core Test Bed specified by 3GPP.
- Design and deployment of low sized IMS Client of which the architecture and specifications are based on the 3GPP, IETF and ETSI-TISPAN standardization bodies has been developed. This IMS Client is targeted at power and memory limited devices and can easily be deployed within mobile devices.
- As a result, this work will contribute to the standardization of service discovery in Personal Networks, providing Instant Messaging and Presence service over the Test bed for pursuing Modeling phase parallel Load Balancing and traffic analysis performed over Test Bed.
- Performing the path ahead with own application creation and deployment over IMS, the MoBlog service client designed and rendering over the IMS Test bed, in real time observation made to Modeling this service as well.

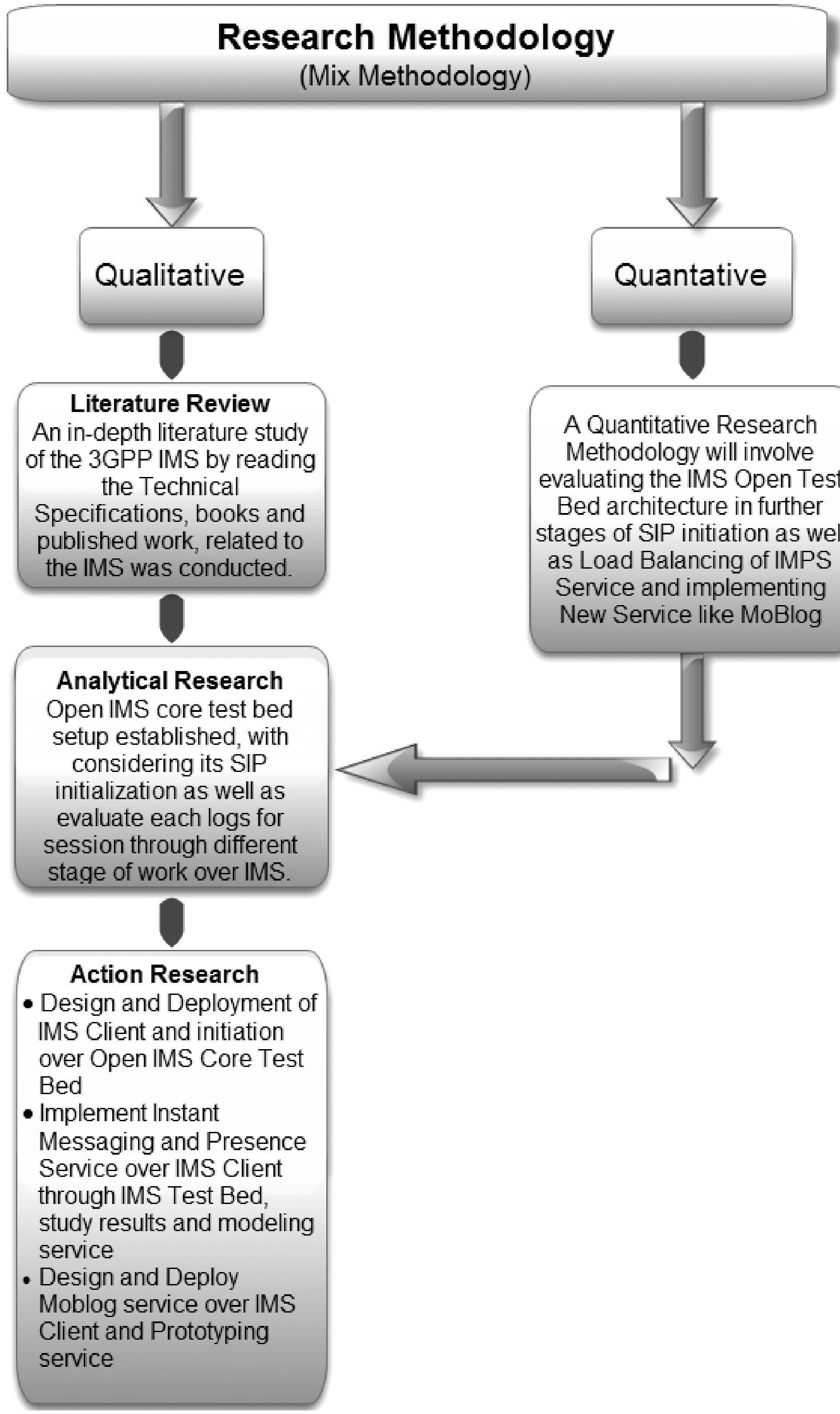

Figure 1.7: Research Methodology for this Research work

1.10 Structure of Thesis

Today with IMS becoming a fast reality, service providers are competing to offer IP multimedia services. Service providers are streamlining their operations and are focusing on efficient network management to facilitate an improved revenue stream by efficient use of network resources. The resulting saving can be used to fund new and improved service options.

The Subsequent chapters of this thesis are organized as following:

Chapter 2: IP Multimedia Subsystem Architecture, Related Technologies & Tools

Chapter 2 describes Requirement for IMS Infrastructure, IMS entities and supportive technologies. This chapter also discusses brief summary of relative technologies for IMS, with IMS Hardware and Software requirement, and IMS tool kit sketch.

Chapter 3: Literature Review

Literature study discusses earlier work and present works in progress, with IMS a brief comparison of all existing researches. With IMS, formed 3 aspects of further research, extending this research from current researcher's problem solving approach and implementation studies.

Chapter 4: OpenIMS Implementation

This chapter represents IMS Testbed setup with Open IMS Core. All tasks related to Test bed setup including each entity configuration and simulation presented here. Also Functionality as well as performance tuning also measured for Open IMS.

Chapter 5: Initialization of SIP IN IMS

IMS relies on the session initiation protocol (SIP) for the development and initiation of applications and services. This chapter discusses SIP elements and functionality. During this research Test bed setup Sip Pre-setup as well as SIP session flow procedure in IMS presented in this chapter, SIP Message captured Session test period also presented with its performance evaluation as part of Open IMS Test bed working phase assessment.

Chapter 6: IMS Client Development and Deployment over Test Bed

After initialized Open IMS Core Testbed and testing with SIP initialization, this chapter discusses IMS Client design and deployment process. In this chapter study of all existing IMS client with limitation to include in this project mentioned. Discuss of all tools used for Client designing and implementation is also mention in this chapter. Main part of this chapter is IMS Client development and implementation in Open IMS Test bed. This phase of research demonstrate Experimental results of IMS client and brief of result discussion.

Chapter 7: Study & Modeling Instant Messaging and Presence over IMS

With this chapter Instant Messaging and Presence service deployment and Architecture details of IMS demonstrated. These works also mention implementation strategy for IMPS server over Open IMS Testbed. Results of various session establishments. Modeling over multiple nodes investigates for IMPS Traffic Analysis as well as Load balancing discusses in this chapter.

Chapter 8: Designing, Deploying & Modeling MoBlog over IMS

Multimedia enriched MoBlog service basics from service over IMS discusses in this chapter. MoBlog able to retrieve, store and share Image, Text and Video from user equipment to IMS network connect other User Equipments (UEs). This chapter demonstrates creation to deployment of MoBlog over IMS and practical functioning phases are discusses with this chapter.

Chapter 9: Conclusion and Future work

Conclusion depicts all research summary as well as result discussion. The aspects conducted during Literature review with brief discussion of assessment of whole work. The future work indicates proposed work for the IMS and Future Web enable communication services with additional service management aspects.