# IMS - Mobile Server Platform
# The foundation of
# Mobile-to-Mobile service networks
# for future cellular Systems

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*"He grants wisdom to whom He pleases, and whoever is granted wisdom, he indeed is given a great good and none but men of understanding mind."*

Quran: Chapter 2, Verse 269

# Abstract

The unprecedented growth of mobile application market is the evidence of twofold technological advancement in the wireless world: first, the enormous increase in the capacity of mobile networks and second, the rapid increase in the computing powers of mobile devices. The former has enabled the network operators to ensure quality of service on their network by inducing more capacity for seamless data transmissions. Whereas, the latter has contributed in the novel space of mobile server paradigm, where the mobile devices are assumed to have sufficient computing power of hosting and distributing small and medium-sized data services among the peers on the network. Nevertheless, considering the mobile server paradigm or peer-to-peer mobile applications, the availability of such applications are mostly limited to WiFi and Wireless Local Area Networks (WLAN) only and, hence, not available for cellular data networks. There are various technical and political reasons behind this phenomena.

The dissertation deals with the standardized provisioning of mobile applications on cellular data networks and addresses the issues related to it. A novel concept of Mobile-to-Mobile service networks is presented in this regard and a prototypical implementation of an IMS (IP Multimedia Subsystem) interfaced Mobile Server Platform (MSP) is provided to enable multimedia communication among the peers. The IMS was originally introduced by 3GPP to support heavy multimedia communication over 3G cellular network, which has now become an application layer standard for 4G / LTE-Evolved Packet Core network.

From the architectural point of view, the MSP is mainly categorized into three different logical frameworks. First, the *Multimedia Messaging Framework* is introduced in order to realize the mobile-to-mobile text messaging and multimedia file sharing features in a service oriented fashion over the operator IMS network. The proposed framework conforms to the functional and non-functional requirements of the Instant Messaging (IM) service addressed by 3GPP.

Second, the *Multimedia Streaming Framework* is introduced in order to support the mobile-to-mobile multimedia sessions in a service oriented fashion over the operator IMS network. The SIP and SDP are the main protocols used for the session establishment process of the streaming applications, whereas, the RTP and RTSP protocols are proposed to provision standardized multimedia transmissions and achieve remote control streaming features on top.

Third, a comprehensive *QoS Framework* is introduced to provide an efficient control of ongoing multimedia sessions according to the Policy and Charging Control (PCC) mechanism devised by the 3GPP. The performance evaluation and mathematical models of File Transfer Time (FTT) and Video Streaming Time (VST) are presented for different network conditions and for various types of user profiles.

# Zusammenfassung

Das beispiellose Wachstum mobiler Anwendungen ist Ausdruck zweier zentraler Fortschritte in der drahtlosen Mobilkommunikation: des enormen Zuwachses in der Kapazität mobiler Netze der dritten und vierten Generation, und dem raschen Zuwachs an Rechnerkapazität und Energieeffizienz mobiler Endgeräte. Die erstgenannte Entwicklung erlaubt Netzbetreibern bruchlose Datendienste hoher Qualität anzubieten, letztere Entwicklung gestattet sogar die Nutzung mobiler Endgeräte als Mobile Service Provider in Peer-to-Peer-Netzen. Allerdings stellt die Kombination beider Effekte die Netzbetreiber heute noch vor erhebliche Probleme, so dass mobile Serviceprovider derzeit schwerpunktmäßig noch auf WiFi und WLAN beschränkt sind. Um dies zu ändern, müssen sowohl technische als auch politische Hürden z.B. im Bereich der Standardeinhaltung überwunden werden.

Die vorliegende Arbeit befasst sich mit der standardkonformen Bereitstellung mobiler Anwendungen in zellularen Netzen, wie sie von den Netzbetreibern heute und in Zukunft angeboten werden. Sie entwickelt ein neuartiges Mobile-zu-Mobile-Servicenetz und fokussiert dabei insbesondere auf die Implementierung von IP Multimedia-Subsystemen (IMS-Standard der 3GPP) für Mobile Serviceprovider, um Multimedia-Kommunikation zwischen Peers zu unterstützen, das nun eine Standard-Anwendungsschicht für das Netz 4G/LTE-Evolved Paket Core geworden ist.

Konkret werden drei logische Rahmenwerke vorgestellt, prototypisch umgesetzt und empirisch evaluiert. Das *Multimedia Messaging Framework* realisiert mobil-zu-mobil Textnachrichten und Multimedia-Dateifreigabefunktionen in einer serviceorientierten Art und Weise über den IMS-Netzwerkbetreiber. Das Framework und seine Umsetzung entsprechen voll den funktionalen und nicht-funktionalen Anforderungen des 3GPP Instant Messaging (IM) -Dienstes.

Das *Multimedia Streaming Framework* unterstützt mobil-zu-mobil Multimedia-Sitzungen in einer serviceorientierten Art und Weise ebenfalls über den IMS-Netzwerkbetreiber. SIP und SDP sind die wichtigsten verwendeten Protokolle für den Sitzungsaufbau der Streaminganwendungen, während die RTP und RTSP-Protokolle standardisierte Multimediaübertragungen und ferngesteuerte Streamingfunktionen in den Griff bekommen.

Drittens wird ein umfassender *QoS Rahmen* eingeführt, um eine effiziente Steuerung der laufenden Multimedia-Sitzungen entsprechend dem 3GPP Mechanismus Policy and Charging Control (PCC) zu erlauben. Auf Messungen sowie ergänzenden mathematischen Modellen beruhende Leistungsbewertungen von File-Transfer Time (FTT) und Video Streaming Time (VST) werden für verschiedene Netzwerkbedingungen und Benutzerprofile durchgeführt.

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*Muzzamil Aziz, Aachen, September 2015.*

# Contents

# Chapter 1

# Introduction

The Mobile-to-Mobile service network is meant to be a network of mobile devices and mobile nodes that are required to collaborate and disseminate any type of multimedia content among each other. The principle of information sharing mechanism in this network is based on the state-of-the-art service oriented paradigm, where every single node of the network is capable of hosting and sharing informational services to other nodes of the network. Thus, considering the exceptional growth of mobile devices and their presence as the major source of information influent society, the demand of such mobile-to-mobile service model is high in every field of life.

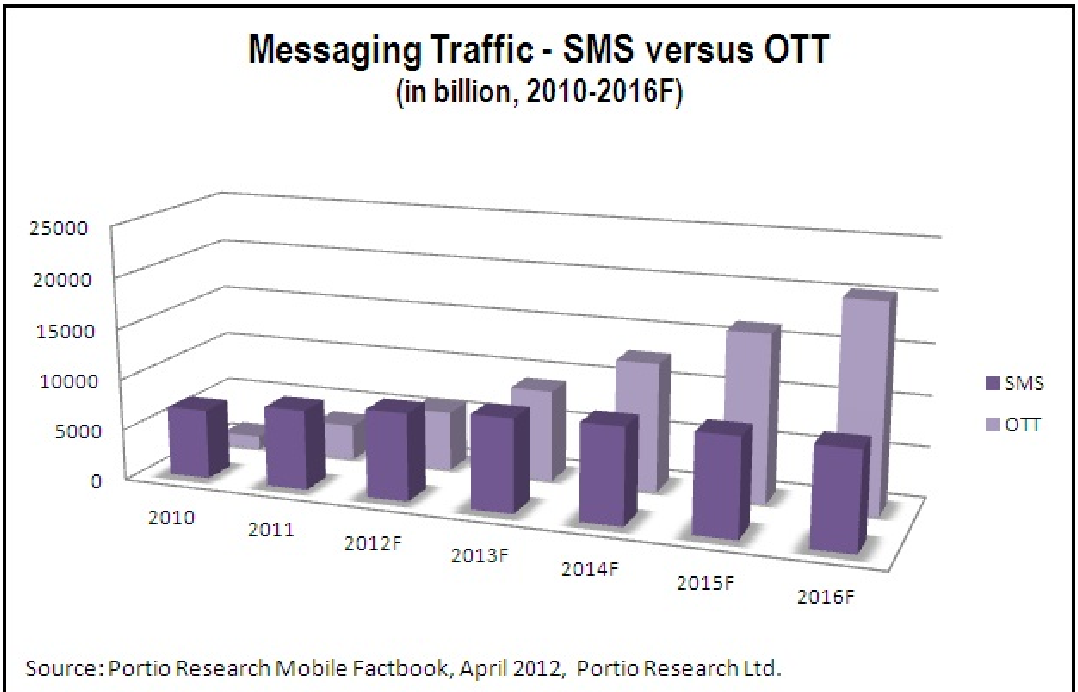
The major upshot of this strategic model is to enrich multimedia communications and informational services among consumer handheld devices. Nevertheless, the similar model is applicable for home appliances, enterprise gadgets and sensor nodes used in home automation, smart metering, vehicle telematics and eHealth applications etc. Here, it is worthy to mention that most of these applications are classified as in-house applications, where a short-ranged communication is sufficient among the devices to operate. Hence, a local area network can be built among devices with the help of bluetooth connections, ad-hoc wireless networks or by creating private WLANs. However, the scope of this dissertation is not limited to in-house applications only but to broaden the space of mobile-to-mobile infrastructure by proposing a standardized service model for mobile cellular networks to support border-less communications between the devices. Such joint venture of IT and Telco systems will bring a plethora of new mobile-to-mobile applications in the market, namely, mobile-to-mobile audio/video messaging and chat, gaming, e-learning, navigation and location applications etc.

From now on in this dissertation, the ”M2M” terminology will be used to specify the concept of ”mobile-to-mobile” communication where ever required. Therefore, the M2M terminology should not be confused with the so called ”machine-to-machine” communication in the context of this dissertation.

## 1.1 Motivation and Objectives

With the epic advances in cellular data networks, the mobile world sees a big market and competition in IP telephony, instant messaging and social networking applications for smart phone users. WhatsApp, Viber, SnapChat, Kik, Facebook mobile and Skype are some major applications that fall in this category [65]. From the cellular operator point-of-view, such applications are collectively known as OTT (over-the-top) applications. “An OTT application is an application or service that provides a product over the Internet and bypasses traditional distribution” [56], that means these applications are offered by third-party service providers by not using the traditional means of billing and distribution channels of the operator network. This leads to a wide-ranging conflict between the telco operators and the service providers who misuse the operator network and offer cheap services on top to the network users.

Figure 1.1 depicts the results of a research study [72] by Portio Research, which compares how smart phone users consume the OTT instant messaging applications versus the legacy telco SMS service. One can see how widely the consumer interest is shifting towards such OTT applications in future. This is a real threat to the traditional telco business ecosystem. Although such OTT applications generate much multimedia traffic over the operator network which seems to be a benefit for the operators, the revenue they generate from the data contracts is much less compared to the extra load over the network by these applications. Therefore, they require somehow a sophisticated control over this traffic that enables them either to block it or to charge it properly as per load to the network. Some architectural advances are proposed by 3GPP (The 3rd Generation Partnership Project) in this




| Traffic (in billion) | 2010 | 2011 | 2012F | 2013F | 2014F | 2015F | 2016F |
|---|---|---|---|---|---|---|---|
| SMS traffic | 6,834.5 | 7,843.7 | 8,599.7 | 9,162.9 | 9,443.5 | 9,639.3 | 9,554.2 |
| OTT traffic | 1,297.1 | 3,491.6 | 5,846.2 | 8,993.7 | 12,682.1 | 16,426.1 | 20,293.1 |

FIGURE 1.1: User subscription of OTT instant messaging vs operator SMS service [63]

regard to the operator's 3G/4G data networks that allow them to install their own application servers and provide their own IP/data services over their network. Hence, the overall extended system will enable the telco operators to own such data services and charge them appropriately in order to take full advantage of this new business model.

The motivation of this research work comes with the aim to utilize the real strength of this new service-oriented business ecosystem. It proposes the foundation of an operator-owned M2M service network which will offer data services to their users, such as instant messaging, multimedia audio/video sharing, IP telephony and conference calling etc. The system will enable the telco operators to control which user is using which service and charge accordingly. Apart from the operators' perspective, the proposed system will also facilitate the smart phone users to have multiple benefits over the OTT solutions. For instance, they can have all the services under a single login to their former telco operator network and do not have to register and pay to several third-party providers for individual applications. A more vigilant and important privilege what they can have is the Quality-of-service (QoS) guarantees from the telco operators. This is the real advantage which OTT providers cannot provide to their customers because they are unable to control the network itself. On the other hand, the operators can have full control over the network resources, and hence, are able to apply different QoS schemes and charging policies for different types of customers and services.

Besides the business point of view, the long distance wireless network coverage is a key requirement for any wide area M2M service deployment. Let's take an example of smart city network as shown in figure 1.2, where the city key locations, displays, public transport and monitoring devices are aimed to collaborate and disseminate information in order to leverage the quality of life index of the city. A cellular network coverage is critical in this setup considering a large area network of both static and highly movable network nodes. Some other potential examples of M2M applications are: Smart metering networks, Car2Car service networks and automated alarm and surveillance networks.

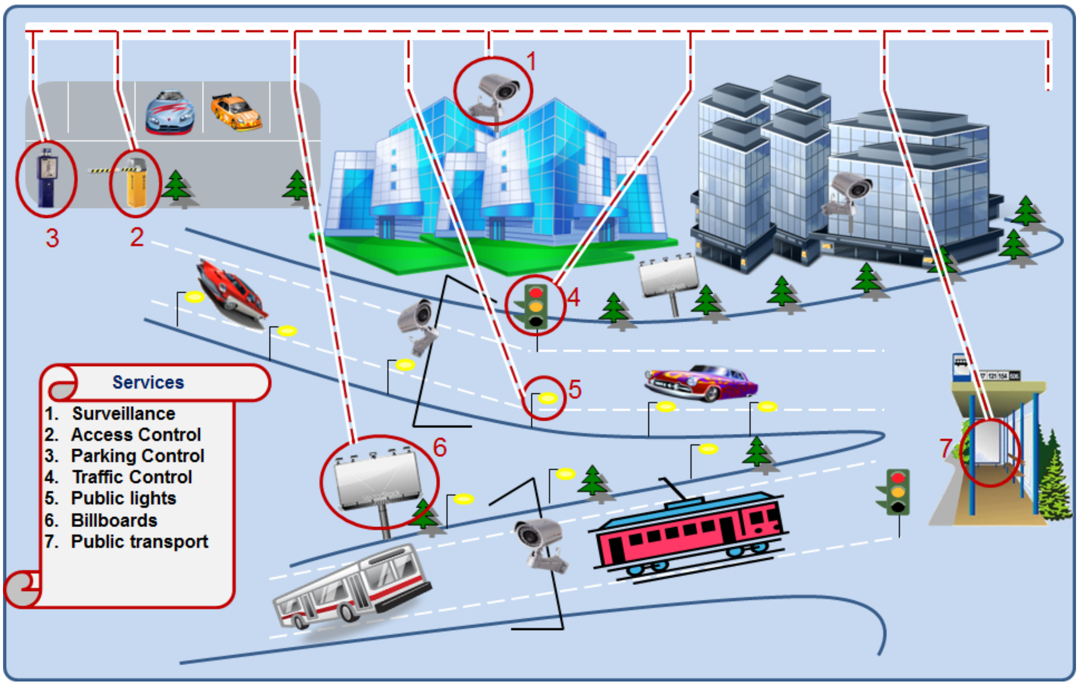



FIGURE 1.2: M2M smart city network concept

Thus, in the investigation of standardized implementation of M2M service network and prestigious M2M services, the dissertation has much room for research and to apply QoS engineering due to some new and complex network technologies in the LTE and IMS system. The real target of this research is to investigate how to efficiently utilize the cellular network resources in order to enhance the overall multimedia streaming experience comparable to the high-speed broadband networks (cable, DSL).

## 1.2 Dissertation Contribution

This dissertation is an extension of the research work by Aijaz[9] and Gehlen[52], which mainly lays the architectural design of a mobile server platform and P2P mobile web services. Without focusing much from the cellular network perspective, the former research was aimed for wireless networks in general, such as WLAN, WiFi networks. However, the major contribution of this dissertation is to bring the existing knowledge of the mobile service layer research to practice into the cellular network world. Following is a list of some key contributions of this research work :

- The dissertation introduces a novel concept of Mobile Application Servers (Mob-ASs) in the operator LTE / IMS network running on mobile devices. Like other Application Servers in the IMS network, a MobAS has the ability to register itself as a mobile host through standardized interfaces proposed by 3GPP.
- Based on the Mob-AS platform, the dissertation implements a M2M service network for LTE-Evolved Packet Core (LTE-EPC).
- The M2M services like multimedia instant messaging and real-time audio video streaming are developed for Mob-AS platform, which are fully compatible to work with Ericsson IMS and Fraunhofer OpenIMS, OpenEPC testbeds.
- In order to address the QoS concerns, the dissertation introduces a QoS framework for Mob-AS platform that fulfills the requirements of 3GPP Policy and Charging Control (PCC) mechanism.
- The dissertation extensively evaluates the behavior of the system based on user classifications and service types.

## 1.3 Structure of Dissertation

**Chapter 2** provides a brief history of the cellular data networks and their services. Without focusing on complex network and architectural details, the chapter gives a quick overview how evolution took place in the packet core domain of a mobile cellular system that enabled it to evolve from a traditional circuit switching to the all-IP network.

**Chapter 3** introduces a novel concept of Mobile Application Server as a foundation of M2M service networks in the cellular world. The chapter first presents the potential challenges that can be faced while realizing an M2M service network over a cellular operator network. Then, it provides the details how an existing Mobile Server Platform can be extended and utilized to enable M2M communications over the operator IMS/LTE network.

**Chapter 4** enfolds the design and implementation of M2M messaging framework of the Mobile Server Platform. The chapter describes how M2M text messaging and multimedia file sharing is realized in a service oriented fashion over the operator IMS network. Furthermore, the chapter provides the performance evaluation of the newly implemented services.

**Chapter 5** presents the design and implementation of M2M multimedia streaming framework of the Mobile Server Platform. The chapter describes how M2M multimedia messaging streaming is realized in a service oriented fashion over the operator IMS network. Furthermore, the chapter provides the performance evaluation of the newly implemented services.

**Chapter 6** presents the QoS framework of the Mobile Server Platform. The chapter enfolds how the proposed QoS framework operates in user registration, session establishment and multimedia delivery phase to provide an end-to-end QoS control. A promising feature of the proposed framework is to allow the cellular operators to employ QoS control based on the user profiles classification. Hence, a systematic approach is adopted to study the behavior of a video streaming prototype based on different network settings and various user profiles.

**Chapter 7** summarizes the complete dissertation work and presents a conclusion at the end.

# Chapter 2

# Evolution of cellular data networks

The mobile phones are not associated with communication only; in fact they are performing many more everyday jobs other than communication. However, the history of mobile phones is quiet fascinating. The basic telephony was first introduced in 1876 by Alexandar Graham Bell. Equipped with basic telephony, the early phones were large and heavy in size as one's forearms and some even came with large antenna like a cordless phone. At that time, the phones were just like two-way radios that allowed people to communicate for emergency services only.

The history of wireless phones starts back from 1908 when a US Patent was presented in Kentucky for wireless telephones. In 1940, the engineers at AT&T first time introduced the concept of division of a wider base station area into smaller cells. In early planning, powerful base stations were utilized to cover a wide area without dividing into smaller cells. Then an era of smart phones appeared in 1993 when the first public smart phone was introduced by IBM and BellSouth. They are equipped with loads of attractive features and capabilities as well as ultra-thin and tech-savvy as compared to their predecessors.

This chapter provides a brief history of the cellular data networks and their services. Without focusing on complex network and architectural details, the chapter gives a quick overview how evolution took place in the packet core domain of a cellular system, which enabled it to evolve from a traditional circuit switching to the all-IP network, as shown in figure 2.1.

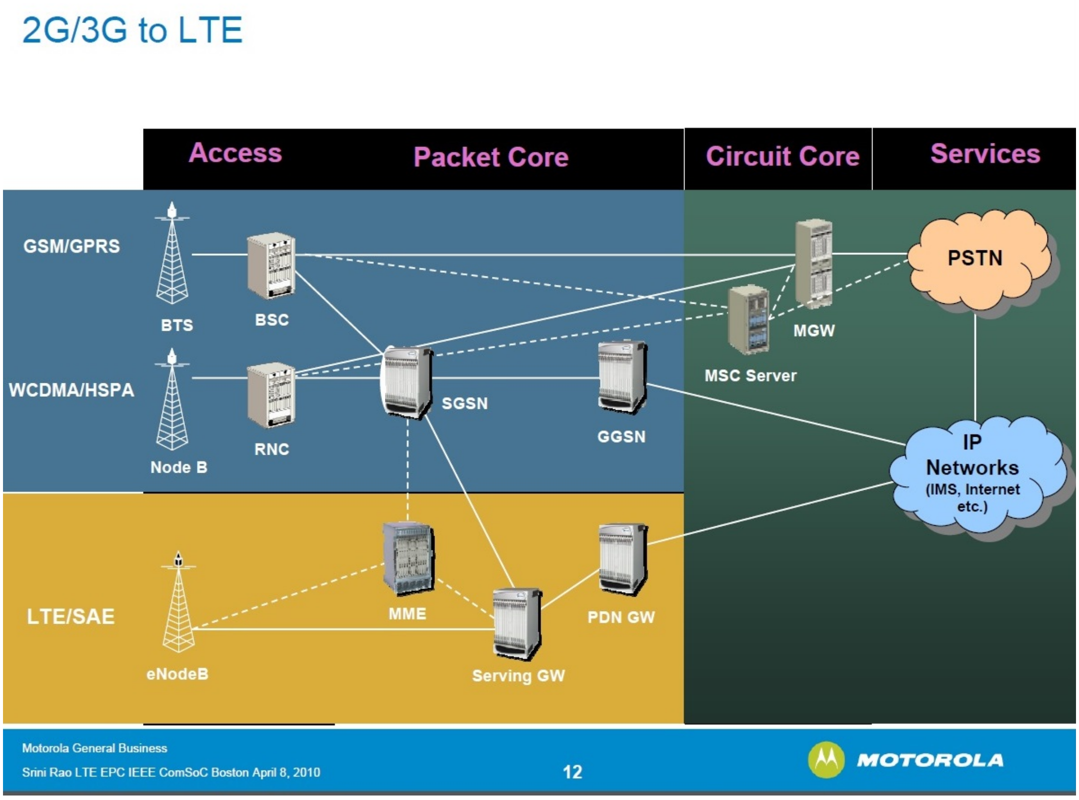



FIGURE 2.1: Mobile network evolution from 2G/3G to LTE[70]

## 2.1 History of Cellular Networks

### 2.1.1 Systems with analogue transmission (1G)

In mobile telecommunications world, the term 1G refers to the first generation of wireless telephone. It is often represented as 1-G. The 1G telecommunication standards were first published in the 1980s [27]. These standards were followed by the telco industry as a guideline until they were replaced by 2G digital telecommunications.

The older cellular networks were designed based on the principle of analogue transmission, such as the first public radio network, the A-Netz, was introduced in 1958 in Germany. Later on, in 1972 and 1986 the A-Netz was followed by the cellular B (restricted to telephony) and C networks (for data services also) respectively [78].

Table 2.1 presents some other details of these networks and a comparison with other analogue systems deployed in Scandinavia (Scand), Great Britain (GB) and USA.

In 1979, Nippon Telegraph and Telephone (NTT) launched an automated cellular network, 1-G in Japan [27]. This was a first commercial project by a telecommunication authority at that time. Likewise, the first 1G project was started in 1983 by Ameritech in Chicago. The first ever mobile phone of Motorala's DynaTac series was used in this project. Figure 2.2 depicts the first portable cellular phone by Motorola that was approved by Federal Communications Commission (FCC) on September 21 1983.

In 1990, numerous eminent cell phones such as the Motorola DynaTAc Analog AMPS were ultimately outdated and replaced by Digital AMPS (D-AMPS). Since the service was outdated therefore it resulted into the down fall and by 2008, AMPS service was shut down by most of the American carriers.

| **Parameter** | **C450** | **NMT450** | **NMT900** | **TACS** | **E-TACS** | **AMPS** |
|---|---|---|---|---|---|---|
| Country | Germany | Scand | Scand | GB | GB | USA |
| Standard | DBP Telekom | | | CRAG | CRAG | FCC |
| Introduced in | 1985 | 1981 | 1986 | 1984 | | 1983 |
| Uplink $MHz$ | 450.3-454.74 | 453-457.5 | 890-915 | 890-915 | 872-905 | 824-849 |
| Downlink $MHz$ | 461.3-465.74 | 463-467.5 | 935-960 | 935-960 | 917-950 | 869-894 |
| Channel spacing $KHz$ | 20 | 25 20 | 25 12.5 | 25 | 25 | 30 |
| Duplex range $MHz$ | 11 | 10 | 45 | 45 | 45 | 45 |
| Access method | FDMA | FDMA | FDMA | FDMA | FDMA | FDMA |
| Modulation | FSK | FFSK | FFSK | PSK | PSK | PSK |
| MAH | Yes | No | No | No | No | No |
| Cell diameter $Km$ | | 15-40 | 2-20 | | | |
| Frequencies | 222 | 180 220 | 1000 | 1000 | 1320 | 833 |
| Data services $kbits/s$ | 2.4 | | | | | 2.4 |
| Traffic capacity $Erl/km^2$ | | | | 14 | 14 | 12 |

TABLE 2.1: An overview of analogue cellular mobile radio [78]

FIGURE 2.2: Motorola DynaTAC 8000X [66]

### 2.1.2 Digital cellular systems (2G)

The era of 90s saw the first digital network rising in order to enhance the sound quality, network security and capacity. The first GSM network was commercially launched by a Finnish GSM operator, named as Radiolinja, in 1991 [59]. Figure 2.3 depicts the first GSM phone call made by a former Finnish prime minister, Harri Hokeri, in Helsinki using a Nokia mobile phone.

These networks initially supported circuit switched data (CSD) technology to attain better performance. The CSD was a mechanism to place a dial-up data call digitally that increased the data transfer rate up to 14.4 Kbps, which was comparable to the transfer speed of an early-to-mid-nineties landline modem. Still, these networks were unable to provide intrinsic and tightly coupled support for data services. The technology of GSM-900 and GSM-1800 are the most popular being used on the globe - Europe, Middle East, Africa, Australia, Oceania (and most of Asia). While on the contrary, the Southern and central parts of America are up using the following technologies [32]:

FIGURE 2.3: First GSM phone call [59]

- Argentina GSM-1900,
- Belize GSM-1900,
- Bolivia GSM-1900,
- Brazil GSM-1800,
- Colombia GSM-800/1900,
- Costa Rica GSM-1800,
- Dominican Republic GSM-1900,
- Paraguay GSM-1900,
- Peru GSM-1900,
- Venezuela GSM-900.

Although the digital networks took precedence with better sound quality, a complete knock-out of analog networks is somehow unfair. Both strategies have their advantages and disadvantages in certain conditions. For example, in analogue networks the call dropping probability is somewhat gradual and partial, whereas dropouts in digital networks are quite sudden and frequent in bad network conditions. In addition to the GSM protocol, 2G also utilizes various other digital protocols, including CDMA (Code division multiple access), TDMA (Time division multiple access), iDEN (Integrated Digital Enhanced Network) and PDC (Personal Digital Cellular). GSM is based on TDMA/FDMA (Frequency division multiple access).

### 2.1.3 GPRS (2.5G)

The GPRS (General Packet Radio Service) marked a breaking point in cellular history when it was launched in 2001 as it offered a continual data services over Packet-Switched (PS) extension of GSM networks [79]. With proper configuration of an account and phone, one could seamlessly use data whenever and however needed. This advancement removed the requirement of outdated dial-up CSD.

GPRS specification was officially released by 3GPP first in Release 97 and later in Release 98 and 99 [1]. GPRS introduced two major network nodes in the traditional GSM network in order to support packet switching in the cellular world, as shown in figure 2.4:

- GGSN - Acts as a gateway which allows the mobile users to access the public data network, such as Internet.
- SGSN - Communicates between the mobile phone and maintains information about the location of the mobile user.

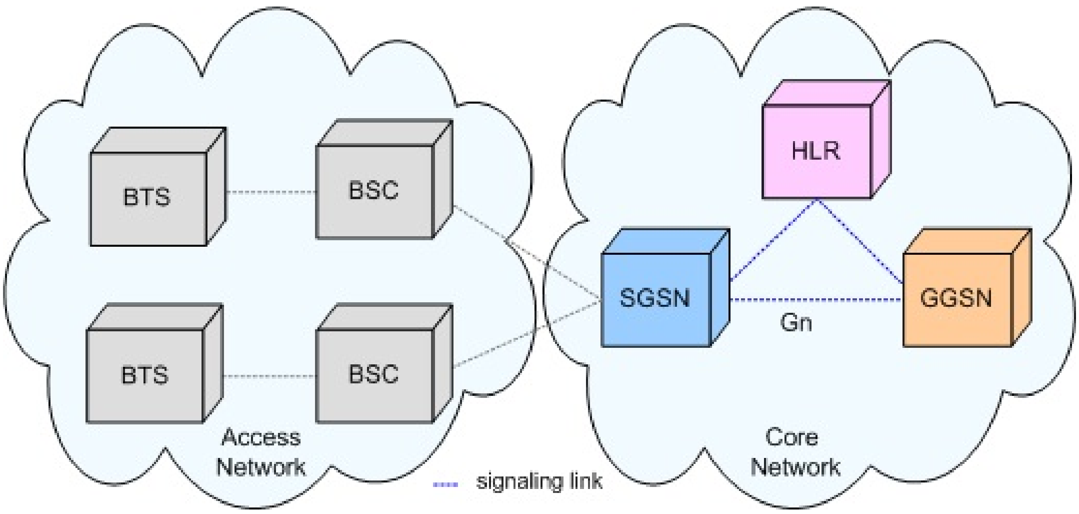



FIGURE 2.4: A typical GPRS network [73]

The service was envisaged to offer faster data rates than GSM, with a theoretical downlink speed of up to 171 Kbps [42]. The innovation of GPRS came at the right time when people really started checking their e-mails regularly. It was a transformation which was warmly greeted by the people all over the globe. The major players in the US market like AT&T Wireless, Cingular, and Voice Stream (later T-Mobile USA) and every major GSM operator in the world, deployed the service. However, this breakthrough in GSM technology did not provide a generational notch. Soon after GPRS launch, the UN's International Telecommunications Union put together its IMT-2000 standard which specified a true 3G technology, requiring stationary speeds of 2 Mbps and mobile speeds of 384 Kbps [83]. Those specifications were tough to meet by GPRS as it couldn't meet these benchmarks even on its best day.

Subsequently, 2.5G technology was used to bridge between 2G and 3G wireless technologies. 2G is primarily referred to describe those evolved technologies that were first considered as being 2G. Unlike 2G and 3G, which have been officially defined as wireless standards by the International Telecommunication Union (ITU), 2.5G has not been recognized by ITU and thus was created only for the purpose of marketing. This is how GPRS got sandwiched between 2G and 3G technology as it was better than 2G but not good enough to be called as 3G. Had ITU not specified 3G, 2.5G might have earned the right to be called as 3G. But now, it seems to just be the first of many generational schemes over the next decade.

### 2.1.4 EDGE (2.5/2.7G)

EDGE also referred as 2.75G is a data connectivity service faster than GPRS (sometimes called 2.5G) but slower than 3G networks. The fact of the matter is that the EDGE has been officially recognized as 3G technology. It is pertinent to mention here that the typical EDGE implementations generally do not obtain 3G data rates, thus, leading people to call it 2.75G. As compared to ordinary GSM/GPRS connection, EDGE delivers threefold increase in capacity and performance due to its higher bit-rates per radio channel through sophisticated methods of coding and transmitting data.

With evolved EDGE continuation in Release 7 of the 3GPP standard, it provides a reduced latency and more than doubled performance to complement High-Speed Packet Access (HSPA). Peak bit-rates of up to 1 Mbps and typical bit-rates of 400 Kbps can be expected. EDGE enhancement of circuit switched data communication is called ECSD (Enhanced Circuit Switched Data). The purpose of ECSD is to increase data transmission rates and to update the existing applications such as high-speed circuit switched data (HSCSD) for EDGE modulation in the radio interface [68]. EDGE enhancement of packet switched data communication is called EGPRS (Enhanced GPRS). BTS and MS devices must be EDGE capable in GSM/GPRS system architecture. For the same purpose, the BTS units must be enhanced with EDGE Transceiver Units (EDGE TRU) due to the higher data communication rates. Only in case of BSS, GPRS and EGPRS have different protocols and behavior.

In Figure 2.5, user plane protocol architecture is depicted where EGDE modified protocols are BSSGP, RLC, MAC and GSM RF. The rest of the protocols are not influenced by the introduction of EDGE.

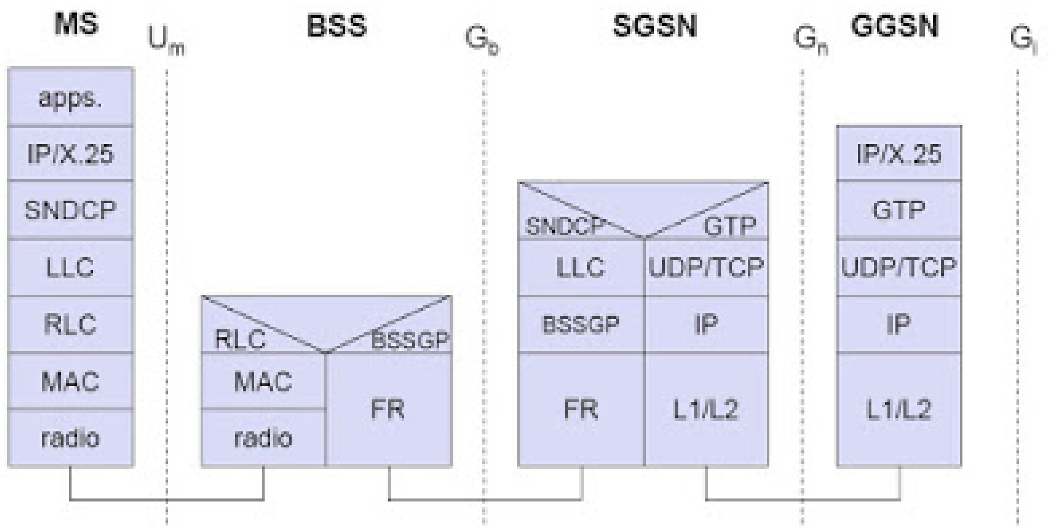



FIGURE 2.5: Protocols introduced by EDGE in GPRS [33]

### 2.1.5 UMTS (3G)

The term (UMTS) refers to Universal Mobile Telecommunications System and likewise term 3G refers to the third-generation in mobile telecommunications world.

It is a third generation mobile cellular system for networks based on the GSM standard. 3G mobile communications standards allow mobile phones, hand-held devices, and computers to access the Internet wirelessly. 3G provides a wider range of services and advances network capacity over earlier 2G networks. It also boosts the rate of information transfer known as spectral efficiency. The telephony has received a wider area and more range, while video and broadband data transfers have also been affected much faster.

It is more complicated to forecast the expected usage of different services for different types of traffic. Video requires high data usage in comparison with E-mail. For example in real-time voice call, audio streaming requires high data usage. It is just like providing dedicated channel for circuit-switched voice but with much greater bandwidth demands.

The first 3G network offered for commercial use was successfully launched in Japan and South Korea [76]. The initial commercial launch of 3G was also done by NTT DoCoMo in Japan. Subsequently, British Telecom in the UK and Monet Mobile Networks in US followed suit. Most countries had implemented the 3G technology by 2007.

3G is about four times faster than the old 2G standards. With initial speed of around 200 Kbps and steady transformation of the technological innovations saw maximum speed of up to 7.2 Mbps. The download speed of 3G network is about 14.4 MB/s and upload speed is 5.8 Mbps. The minimum speed for a stationary user is 2 Mbps, whereas a user in a moving vehicle can expect 348 Kbps. Some distinctive services associated with 3G are: Voice telephony, Broadband wireless data, Mobile video, Mobile e-commerce, Location-based services, Mobile gaming and Audio-on-demand.

The 3G network uses a different frequency than 2G. This eventually forced many operators to build entirely a new infrastructure and obtain additional licenses. Based on these reasons, countries like China and Indonesia deliberately chose to hold back the network from its citizens for many years.

### 2.1.6 LTE (4G)

LTE is a short form for Long-Term Evolution, which is usually known as 4G LTE. It is a radio and wireless broadband technology to support high-speed data for mobile phones and handheld devices. 4G LTE allows operators to attain higher peak throughputs than HSPA+ in higher spectrum bandwidth.

In 2004, 3GPP initiated work on LTE and a completed 3GPP Release 8 in March 2009 [17]. The first LTE network was commercially launched in December 2009 by TeliaSonera in Norway. In last few years, 4G LTE has brought epic advances in the fields of education, health, transportation, and at enterprise level.

4G LTE is one of the several competing 4G standards that offers Ultra Mobile Broadband (UMB) access to the web from mobile devices such as smart phones, laptop computers with wireless modems. It supports not only quicker access and ultra-fast speed to the web from mobile devices, but also opens up new opportunities for video conferencing, streaming high-definition videos and cloud computing.

The core objective of 4G LTE networks is to sustain higher data rates, high performance radio access technology, abridge the network architecture by utilizing all packets that eventually reduce network latency. Since 4G LTE has a scalable bandwidth, so this allows operators to easily migrate their networks and users from HSPA to LTE over the period of time. LTE is designed in such a manner that supports voice in the packet domain. It integrates the best in a given line of radio techniques to achieve higher performance levels that will not be attainable through with CDMA approaches, essentially in larger channel bandwidths.

In integrated networks, LTE systems coexist with 3G and 2G systems in the same way as 3G coexists with 2G systems. Multi-mode devices will function across LTE/3G or even LTE/3G/2G depending upon market situation. Below mentioned are some unique capabilities of the LTE network: [17]

- Downlink peak data rates up to 326 Mbps with 20 MHz bandwidth.
- Uplink peak data rates up to 86.4 Mbps with 20 MHz bandwidth.
- Operation in both TDD and FDD modes.

- Scalable bandwidth up to 20 MHz, covering 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz, and 20 MHz in the study phase.
- Increased spectral efficiency over Release 6 HSPA by two to four times.
- Reduced latency, up to 10 milliseconds (ms) round-trip times between user equipment and the base station, and to less than 100 ms transition times from inactive to active source.

## 2.2 State of the Art Technologies

The above section provides a general overview of the cellular network technologies used in the past. Hence, this section presents some more specific details of the state of the art tools and technologies that are currently being practiced in the telecom industry and academia as well as relevant to understand the technical details of this research work in particular.

The state of the art discussion is divided into three different streams : 1) Packet core services and solutions based on IMS and LTE ecosystem 2) Mobile Server Platform 3) Quality of Service in Mobile networks

The first stream explains how data services are deployed in the IMS/LTE system and how policy and charging control works for multimedia applications. Then, it gives an overview of the industrial standards and solutions to deploy voice and messaging services over the cellular operator networks using the IMS and related technologies. The second stream discusses the mobile server platform research and presents the state of the art in this regard. The third stream focuses on the QoS control of the multimedia applications deployed in the mobile networks. It highlights the QoS issues and presents the proposed solutions both from academia and industry.

### 2.2.1 LTE - EPC and 3GPP PCC Architecture

System Architecture Evolution, or SAE, is identical with Evolved Packet Core, or EPC. SAE/EPC is defined by 3GPP in Release 8 [7] as a completely new core network with a flatter all-IP architecture. It is a low latency packet-optimized system that supports multiple radio-access technologies, with the assumption that the system will support all services including voice in the packet-switched domain [16]. The radio access technologies supported by EPC include 3GPP networks, such as GERAN (radio access network of GSM/GPRS) and UTRAN (radio acces network of UMTS-based technologies), as well as some non-3GPP networks, such as WiMax, cdma200, WLAN and fixed networks. The non-3GPP networks are further categorized into "trusted" and "untrusted" networks. However, 3GPP leaves this decision to network operators to specify which non-3GPP networks are treated as trusted or untrusted.

The 3GPP's aim is to establish an access-agnostic policy control framework, with the objective of introducing a standard for quality of service and policy mechanisms for multi-vendor deployments which allow the operators to offer service and subscriber differentiation. The standard set by 3GPP elaborates about creating transmission paths between the external packet data network (PDN) and the user equipment (UE) with distinct QoS. Bearer Model has been introduced by 3GPP to implement QoS. A bearer is a logical channel used for a service with well defined QoS parameters.

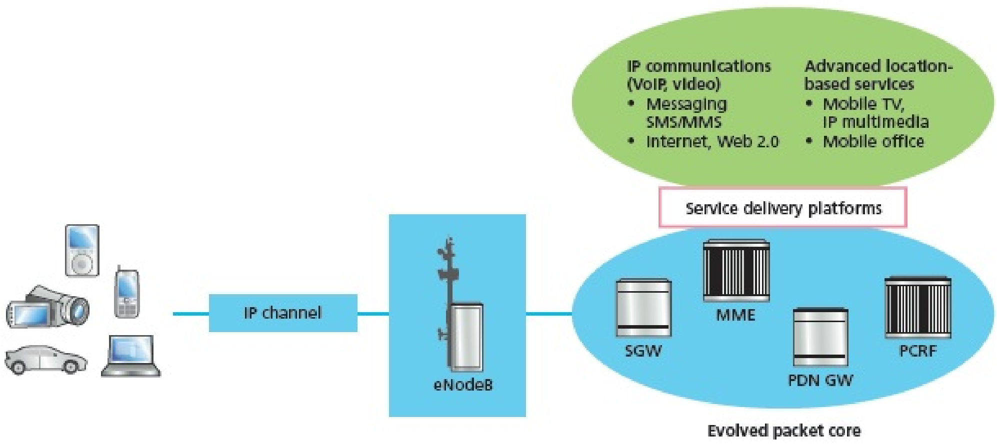



Figure 2.6: LTE - Evolved Packet Core [14]

Figure 2.6 presents a basic architecture of LTE-EPC comprised of various network elements: the Mobility Management Entity (MME), the Serving Gateway (SGW), the Packet Data Network Gateway (PDN-GW) and the Policy controller (PCRF). As depicted, the architecture supports provisioning of IP communications and various multimedia services by connecting to some service delivery platforms (e.g., Application Servers in the IMS).

1. **Mobility Management Entity (MME)** The MME deals with the control plane signaling of LTE EPC architecture. It is responsible for initiating paging and authentication of the UE towards the central user database. It is a termination point of Non Access Strartum (NAS) signaling from the UE, which is used for initial attachment, authentication and service requests etc. Among its other duties, the MME is also responsible for UE authorization to Public Land Mobile Network (PLMN), enabling roaming restrictions and load balancing between SGWs etc.

2. **Serving Gateway (SGW)** The Serving Gateway (SGW) is a network entity that deals with the transportation of IP data traffic between the UE and the external networks. In other words, it is a gateway that serves the UE by routing the incoming and outgoing IP traffic. Moreover, it is responsible for acting as a mobility anchor-point for handovers with the neighbouring eNodeB's, and between LTE and other 3GPP technologies (e.g., 2G/3G systems).

3. **Packet Data Network Gateway (PDN-GW)** The PDN GW (also known as PGW) is the point of interconnect between the EPC and the external packet data network. The UE may simultaneously connect to more than one PDN-GWs to access multiple packet data networks.

4. **Policy and Charging Rules Function (PCRF)** Numerous nodes in the LTE and EPC access play a significant role in implementing QoS and policy management. The policy server (known as PCRF) is the key figure out of them. The service session-level policy decisions are produced with the help of PCRF which tend to work by taking the obtainable network information and operator configured policies. The decisions which are also called the PCC (Policy and Charging) rules are then further sent to the policy and charging enforcement function (PCEF) which is found in the PDN-GW. The PCEF imposes the policy decisions by introducing bearers, mapping the service data flows to the bearers and executing traffic policies. [8]. The PDN-GW maps the bearers against the underlying transport network. The transport network at times uses the MPLS but normally it is based on the Ethernet. The standard IP QoS techniques are used by the transport network as it is ignorant of the bearer concept.

The end-to-end QoS and policy enforcement is managed with the help of eNodeB. eNodeB is the radio base station in LTE. Following are the three main functions of eNodeB: [8]

- Makes the uplink and downlink rate policing
- Defines RF radio resource scheduling
- Uses ARP for the allocation of bearer resources

The radio resource scheduling algorithms in eNodeB have a huge influence on the overall network performance as well as on the quality of service. Mobile network equipment manufacturers (NEMs) have to consider that network operators have a lot of chances to connect their eNodeB products to competitor's products. The UE has a key role to play in the uplink direction. The bearer and service data flows mapping is done with its help.

### 2.2.2 IP Multimedia Subsystem (IMS)

The IMS defines a generic framework for enabling the convergence of voice, video and data communications over IP-based infrastructure using Session Initiation Protocol (SIP) [53]. It was originally introduced to evolve 3G / UMTS data core network by 3GPP (release 5) to enable IP multimedia communications based on widely adopted Internet standards and protocols defined by the Internet Engineering Task Force (IETF), such as SIP, IPv6 and Diameter [44]. Later in subsequent releases 6 and 7, the IMS specification was further refined and maintained as a part of 3G packet core network [2]. In release 8, the IMS is adopted as a application layer standard with the introduction of LTE-Evolved Packet Core (LTE - EPC).

The IMS actively plays its role in the control signaling part of IP multimedia communications, leaving the PS core network to provide transport for the data transmission. In other words, the IMS is not aimed to produce or bring new IP multimedia services in the operator network but to facilitate the provisioning of such services with the necessary QoS control, charging, routing and messaging mechanisms etc [52]. Subsequently, the network operators have more control over the IP services running over their network.

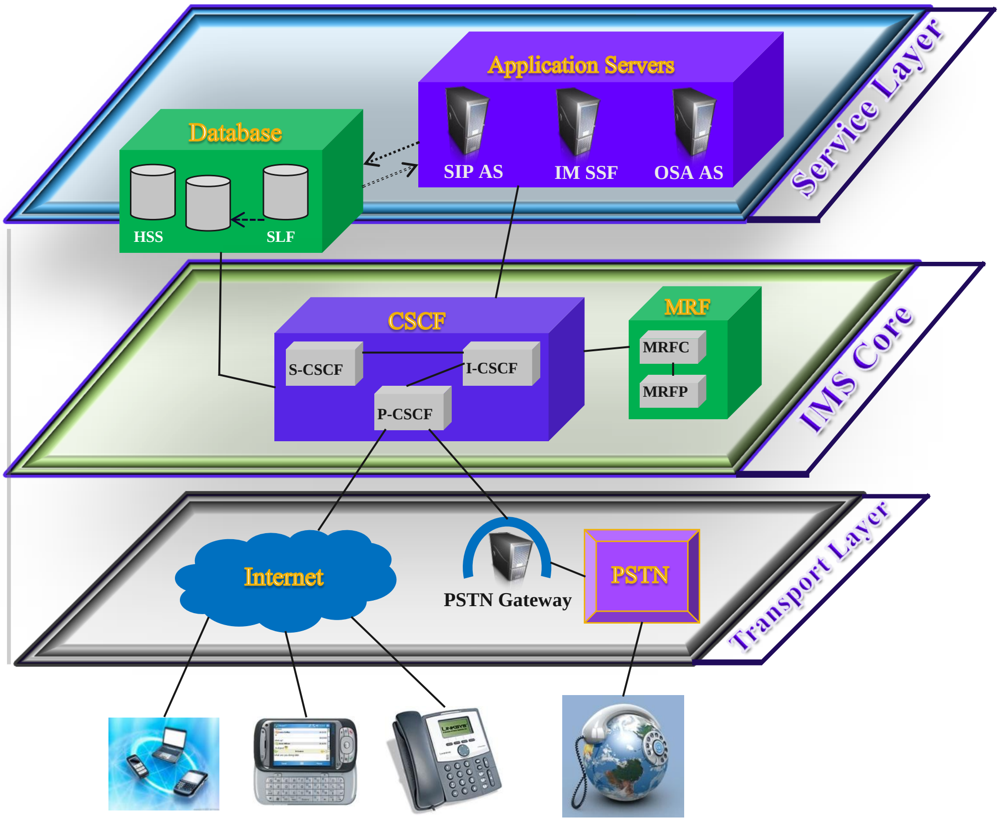



FIGURE 2.7: IP Multimedia Subsystem

A high-level view of 3GPP-IMS ecosystem is shown in Figure 2.7. For simplicity, the whole architecture is divided into three different layers [35, 40]: Transport layer, IMS Core and Service Layer. It is important to note that 3GPP specifies the IMS architecture as a collection of functions not nodes, leaving the choice to vendors to define single or multiple nodes to implement these functions. Figure 2.7 shows that the so-called IMS Core is accessible by any user device connected to the Internet, e.g., via WLAN, ADSL or packet data network such as GPRS using a radio link. A user device capable of communicating to and from IMS Core is typically referred to as an IMS terminal. The transport layer facilitates an IMS terminal to make and receive calls to and from PSTN network or any other circuit-switched network using PSTN/CS gateway as shown in figure.

The IMS Core contains one or more SIP proxies, collectively known as CSCFs(Call/Session Control Functions), MRFs (Media Resource Functions) and user databases, called HSSs (Home Subscriber Servers) and Subscriber Location Functions (SLFs). The CSCF is the most essential part of the IMS Core. It controls all the signaling information to and from IMS terminals and Application Servers(ASs). A CSCF can be one of the three different types [36, 40]: P-CSCF (Proxy-CSCF), I-CSCF (Interrogating-CSCF), S-CSCF (Serving-CSCF).

1. **Proxy Call Session Control Function (P-CSCF)** The P-CSCF claims to provide a reliable transmission of the SIP signalling. It is originally a SIP proxy which acts as the main contact point that is provided to the user terminals by the IMS domain. P-CSCF supports the SIP signal compression and plays a significant role in exposing the IMS Emergency Services.

2. **Interrogating Call Session Control Function (I-CSCF)** It is the principal point of contact in the home network, which acts as a SIP Proxy located at the edge of the network. I-CSCF helps in the discovery of S-CSCF where the user is trying to register. The SIP request is then further directed to its allocated S-CSCF.

3. **Serving Call Session Control Function (S-CSCF)** S-CSCF acts as the kernel of the IMS system. It has the liabilities, such as session maintenance, routing and translation, charging and transmission of information. It also assists in user authentication and in getting the user's service profile.

4. **Home Subscriber Server (HSS)** It is a centralized database containing the information about all IMS users as well as subscribers. Network units handling calls and sessions work on the basis of data provided to them by HSS. If a new user is registered in the IMS domain, the user profile is accessed via HSS and is made available to the CSCF. A Subscriber Location Function (SLF) solves the problem of the location of the subscription data in case of multiple HSS.

5. **Breakout Gateway Control Function (BGCF)** The BGCF is a logical unit which is part of the IMS network. It is responsible for making decisions about the routing of the telephony sessions initiated within the IMS network and which are intended for circuit switched networks (PSTN or other wireless networks).

6. **Media Resource Function (MRF)** The Media Resource Function (MRF), as the name implies, is there for providing media services in the home network. It is responsible for the management and processing of media streams which includes voice, text-to-speech conversion and video etc. Every MRF available

in the network may be provided by a Media Resource Function Controller (MRFC), a signalling node which facilitates the S-CSCF as a SIP User Agent.

7. **Session Border Controller (SBC)** The SBC is also known as the Border Control Function. SBC is an IP-to-IP gateway which is deployed on the border between an operator's own IMS network and other IMS networks. It is a Network to Network interface (NNI). Broadband access to the IMS by UEs can be achieved with the help of the policy enforcement functionality and the P-CSCF; they can team-up to implement such a Session Border Controller which supports the User-to-Network Interface (UNI).

8. **Media Gateway Control Function (MGCF)** It is the vital node of the PSTN gateway. It controls the media resources involved in the flow of traffic between networks having variant types of media. It classically works with an IP based network along with Time Division Multiplexing (TDM) network. Its interaction with the call and session control functions is based on the Session Initiation Protocol (SIP). Also, it collaborates with the control plane of the GSTN using ISUP and nevertheless, with the Media Gateway using the H.248 protocol.

9. **Media Gateway (MGW)** The Media Gateway Control Function (MGCF) controls the Media Gateway (MGW) with the help of H.248 protocol. It helps with the media flows among various networks. It deals with a number of different formats which are there for media transportation including RTP/UDP/IP, TDM and video and voice transcoding.

10. **IMS Application Servers** SIP application server can host several services or it may as well be dedicated for a single service. To provide the user with an unmatched unified user experience, IMS facilitates to combine services from various application servers. Centralization, rapid network access and the simplicity of the application development process are some of the core advantages of the SIP application server. The updates as well as the upgrades are more reliable now due to the centralization of the business logic on fewer SIP application servers. There is no chance of getting an upgrade that belongs to some older version or other similar problems.

### 2.2.3 IMS/LTE based Service Networks

A case study was performed by the Infonetics Research firstly in 2011 and later in 2013 to spotlight the driving forces of the IMS involving the trek of legacy networks to IMS. The table 2.2 depicts the top 3 position holders for the driving factors as well as the percentage of the service providers rating.

| **Motivation Drivers for new Technology** | **% of Service providers rating** definitely a driver |
|---|---|
| Legacy equipment obsolescence | 50% |
| Ability to offer converged services | 47% |
| LTE network deployment | 47% |

Table 2.2: Top 3 drivers for migrating to IMS, statistics 2011 [71]

Infonetics Research conducted another study in June 2013 to name the top driving factors for LTE network deployment like Global standard for network infrastructure and services, the potential to offer converged services and to update the network with the gems of modernistic technology were all part of the theme. The operator percentage reasoning of IMS driver is depicted in the following table 2.3.

| **Motivation Drivers for new Technology** | **% of operators reason a strong driver** |
|---|---|
| LTE network deployment | 72 % |
| Global standard for network infrastructure & services | 53 % |
| Ability to offer converged services | 39 % |
| Modernize network with latest technology | 39 % |

Table 2.3: Top drivers for Operators migrating to IMS, statistics 2013 [71]

IMS and future network convergence set off the noteworthiness of the growing need of IMS for the present and future telecommunication trend. As a result, the vendors have converged their focus on IMS supply chain for the future telecom industry. Heavy Readings held a comprehensive study to evaluate the competency level of the vendors connected with IMS domains. The classification of the company, their IMS strategy and the technology vendors are mentioned in the table 2.4. The table shows that every individual vendor has focus on his particular specialized area and is in competition with his opponent whereas some of the vendors listed in the table carry out other IMS functions as well.

| **Type of Company** | **IMS Strategy** | **Key Technology providers** |
|---|---|---|
| Major Telecom Equipment Manufacturer | Usually covers all aspects of IMS but normally includes partner for some elements development | Alcatel-Lucent, Ericsson, Siemens, Nortel, Nokia, Huawei, ZTE, NEC, Motorola |
| Softswitch Vendors | Leverage VoIP expertise to compete with the major vendors for NGN IMS | Cirpack, Italtel,Sonus, NetCentrex, Veraz |
| Computing and IT Platform Providers | Use IMS to break into proprietary range of areas | BEA, HP, Intel, Sun, IBM, Solid, RadioSys |
| Control-Plane Specialists | Normally focused on innovative ideas | Apertio(HSS), Gallery IP Telephony (CSCF), Leapstone(S-CSCF), VeriSign (HSS) |
| Application-Server Specialists | Use IMS to build stronger position into Telco market | AePona, BridgePort, BroadSoft, NewStep, Persona, Personeta, Ubiquity |
| Media Resource and Gateway Vendors | Best supplier for IMS functions like MRFC or MGCF | AudioCodes, Brooktrout, Convedia, IP Unity |
| Session Border Control or Policy Management Providers | Help in defining policy management according to evolving standards | Acme Packet, NexTone, Operax, Tazz, Bridgewater |
| Client Software Providers | Fill the gap in IMS specifications for handset software | Ecrio, SIPquest, Sonim |
| Signaling Vendors | Build a better link with the legacy networks and IMS | Tekelec, Ulticom |

Table 2.4: Service providers playing in IMS space [45]

The following subsections provide some examples of packet core solutions and products offered by industry and research institutions.The focus of the discussion is the deployment of VoLTE, SMS and Instant Messaging related services.

#### 2.2.3.1 Ericsson VoLTE portfolio

Ericsson is facilitating the telecommunication industry with various types of services, nevertheless is one of the leading vendors of IMS and VoLTE solutions as well. Reports inform that more or less 70% of the mobile network operator's revenue is produced from the voice and SMS services. Also, it is highly anticipated that voice service would be the most treasured service in the LTE and NGN networks.

LTE, RAN, EPC, IMS, other Data management and global services are all part of Ericsson VoLTE solution. It is based on 3GPP standards. Below mentioned categories portray a picture of Ericsson end-to-end VoLTE solution [41]:

1. **Mobile**
   - IMS and Multimedia Telephony Application Server (MTAS)
   - Mobile Softswitch solution
   - Subscriber Data management
2. **Voice support in EPC**
   - **Mobility management entity (SGSN-MME)** is an upgraded version of SGSN for EPC. It supports multi-access, GSM, WCDMA, LTE and interworks with WiFi and CDMA.
   - **Mobile Packet Gateway (GGSN-MPG)** is an additional Gateway entity for large network operators with an installed base of GGSNs, also having a large number of subscribers.
   - PCRF support
3. **Voice support in LTE RAN**
   - Subscriber device

CSFB (Circuit Switched Fallback) for voice services was used by the operators in the initial release of LTE smart phones. The CSFB solution was commercially launched in 21 countries of Asia-pacific, North America and Europe by the end of December 2012. Ericsson is vigorously involved with a chipset vendor for the availability of

VoLTE based phones. SRVCC (Single Radio Voice Call Continuity) solution is also being intensely developed and tested by Ericsson. [41]

**Diameter Signaling Controller (DSC)**

The protocol for the control signaling in LTE and IMS network is known as the Diameter signaling. However, Diameter signaling is controlled and secured using the Ericsson DSC. Following are the advantages of DSC [39]:

- Significant reduction of Operational expenditures
- Robust network signaling
- Less vulnerable to IP based attacks

A number of characteristics of the Ericsson DSC product make it out stand in comparison to the portfolios available in the market. The Diameter protocol triggers the control signaling which is a very crucial feature in the network. User performance related features are very important like authentication of the user, charging and mobility management. They are based on the Diameter protocol and any failure in this channel would affect the end user. The Ericsson DSC is aimed to provide scalability and in-service performance for control signaling products and perform multiple roles, such as: [39]

- **The Diameter Agent (DA)** controls the messaging between the network nodes and is also used to route the signals.
- **Diameter Edge Agent (DEA)** supports the LTE roaming by routing the signaling messaging between the network elements.
- **Diameter Routing Agent (DRA)** routes the signaling message for policy and charging function to the IMS.

The Ericsson DSC addresses a few main scenarios to satisfy the increasing demand of Diameter signaling [43]:

- Centralized routing
- Overload protection
- LTE roaming support

- Session binding
- Address resolution of nodes

#### 2.2.3.2 Fraunhofer FOKUS OpenIMS and OpenEPC testbed

FOKUS competence center NGNI and TU Berlin -AV established a cutting edge testbed system for 3GPP EPC known as the OpenEPC. It will assist not only the industry, but academic institutes (Universities, Research Centers etc.) as well in developing a real-world working scenarios of the 4G technology. The FOKUS team has provided number of updates for this test-bed and the version 4 from May 2013 is under consideration of this research. The interfaces provided in the version 4 have given support for a lot of different access technologies. The OpenEPC toolkit is thought to be one of the most advanced and customizable platform for the fundamental testing of functions of cellular networks [49]. Testing of latest releases of cellular technologies like GSM, UMTS, HSPA and LTE is supported by OpenEPC. Following are some of the advantages of the available solutions as a result of the usage of OpenEPC along with 3GPP LTE Rel.11 [49]:

- The core functional components and reference points specified in 3GPP standard for the base architecture are part of it.
- Modem and phone testing is supported.
- It facilitates the opportunity to test the integration of LTE-EPC off-the-shelf 2G base stations for GPRS/EDGE, 3G base stations for UMTS/HSPA, 4G base stations for LTE/LTE Advanced and Wi-Fi.
- Its key characteristics involve the integration of IP Multimedia System (IMS) (www.openimscore.org) for new Voice-over-LTE (VoLTE) technology.
- The support provided to access the source code under specific licensed conditions.
- All the forthcoming evolutionary concepts like Network Function Virtualization (NFV), Software Defined Networking (SDN) and Self-Organizing Networks (SON) are supported.

The OpenEPC has brought great revolution as a testing environment for mobile network operators to provide support for testing 3GPP IP based systems like LTE, HSPA and EDGE along with Wi-Fi (a non-3GPP based system) [49].

### VoLTE demonstrator using OpenIMS Core and OpenEPC

The IMS is considered to be the most reliable solution for providing multimedia services focusing QoS for the end user, and for the VoLTE network. The open source project of OpenEPC completely supports OpenIMS Core comprising PCRF module for user policy management.

Figure 2.8 depicts the integration of OpenIMS Core with the OpenEPC (PCRF is core component) for multimedia voice service. For instance, a UE registers with the OpenIMS Core framework and starts a multimedia session. The user authorization and the profile inspection will be performed by the OpenIMS Core's P-CSCF node with the help of PCRF. Here, PDN-GW (Packet Data Network Gateway) and AN-GW (Access Network Gateway) represent the respective networks across which the media traffic flows. Additionally, the PDN-GW is responsible to impose the bandwidth requirements for multimedia services as suggested by the PCRF. Hence, the OpenEPC and the OpenIMS system support each other to realize the VoLTE service through the operator owned IMS server. [48].

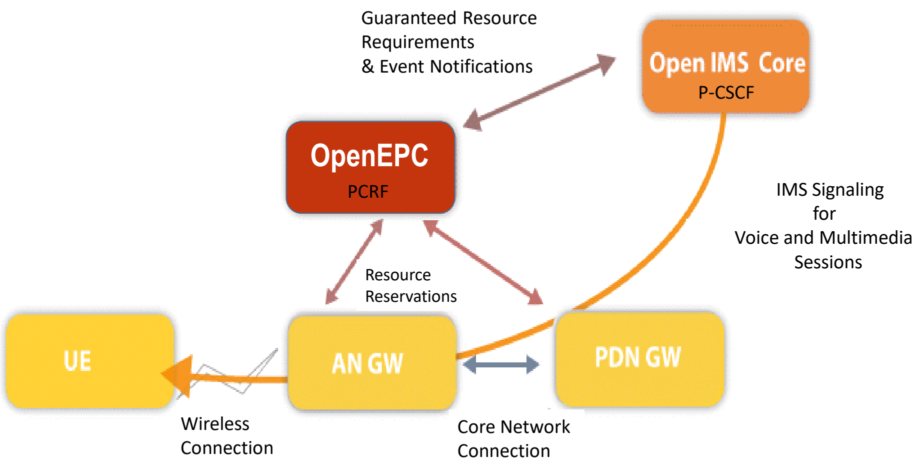



FIGURE 2.8: OpenIMS and OpenEPC integration for multimedia services [48]

#### 2.2.3.3 Alcatel-Lucent Solution for IMS Services

Vendors follow different strategies for reducing operational cost as well as boosting the experience of the end user. Vendors install various applications in their respective networks with the help of Alcatel-Lucent which is considered the market leader in the IMS space additionally providing the end-to-end IMS solution. Besides providing the IMS services to the wireless operators, the Alcatel-Lucent also helps the fixed line providers. AT & T (U.S.) is one of its biggest customers using Alcatel-Lucent IMS services.

**Alcatel-Lucent 4G LTE solution** Along with the end-to-end network management Alcatel-Lucent delivers an IMS solution for the 4G LTE EPC. The outline of the IP based LTE solution which comprises wireless packet core, converged backbone, software defined radio, mobile back haul and the IMS application enablement module is shown in the figure 2.9. This solution qualifies with the open Application Programming Interface (API), which enables the vendor to innovate and enhance the applications stack.

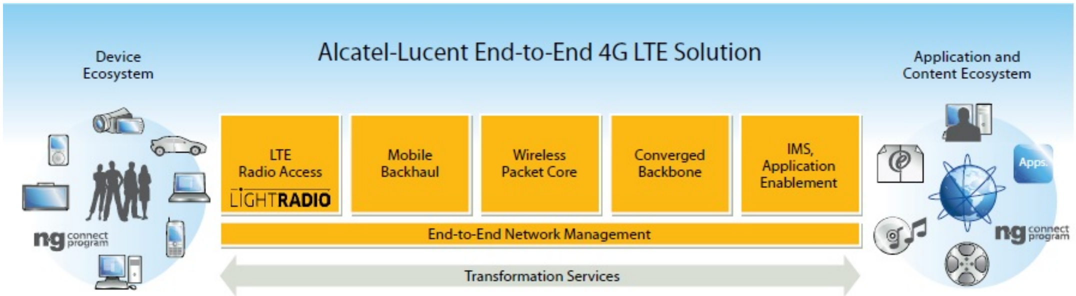



FIGURE 2.9: Alcatel-Lucent End-to-End 4G LTE Solution [64]

#### 2.2.3.4 Huawei LTE-EPC Deployment and Services

Figure 2.10 depicts an instant messaging solution of Huawei LTE network for both IMS and non-IMS scenarios. In a non-IMS scenario (1), the MME is linked to the SMS-IW-MSC (SMS-Interworking-Mobile Switching Centre) which is the part of the CS domain. In scenario (2), the transport level interworking linking is within the IMS domain. Whereas, in the service level networking (3), the IP-SM-GW (IP Short Message Gateway) within the IMS domain takes care of the IM/SMS interworking. [34]

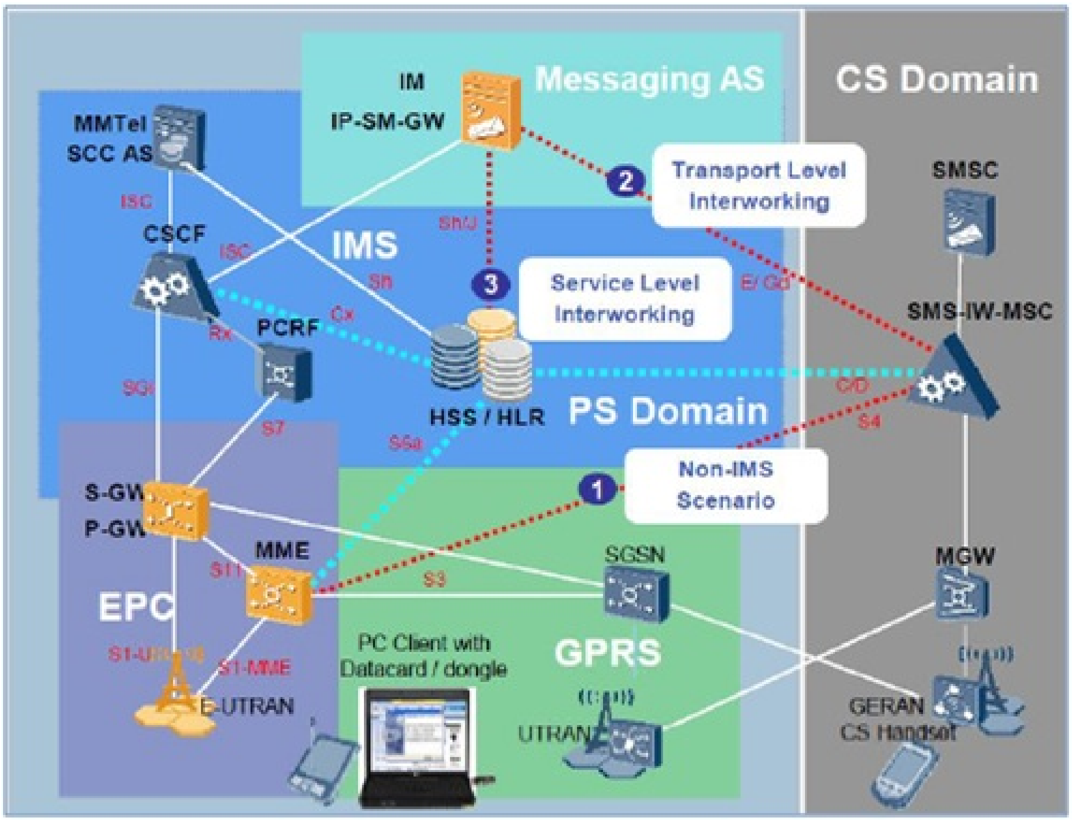



FIGURE 2.10: Huawei - Messaging using IMS over LTE [34]

Last but not the least, the messaging solution proposed by Huawei is in complete compliance with the OMA and 3GPP standards (TS23.204 [6]).

### 2.2.4 Mobile Server Platform

Despite few related works dedicated to propose software architectures for mobile application servers, researchers are more focused in enabling mobile devices as standardized application/service clients. Moreover, concerning operator's telecom network and IMS service layer architecture, there exists no precedence of the concept of mobile application server hosting IMS services proposed for the foundation of mobile-to-mobile service network.

The concept of enabling a mobile device as web services host was first introduced by Guido in [52]. The proposed mobile server framework was capable of successfully hosting short-durational mobile web services suitable for synchronous SOAP communication for a client-server model. Later on, Aijaz presented an extended asynchronous architecture for mobile server framework [9] to support long-durational mobile web services. In their further study, the same research group proposed a new RESTful architecture for their mobile server framework and termed it as light weight compared to the usual SOAP mobile web server [10] [12] [11]. Meanwhile, Srirama also presented the architectural requirements of a SOAP based mobile web server [69]. In his research findings, he claimed that the total processing time of a simple mobile web service on a Mobile Host takes only a small fraction (<10%) of the total request-response time. Whereas, the rest of the response time is because of the transmission delay. For accessing multimedia services from mobile terminals, a research work from academia [30] presents how the Lightweight Application Server (LAS) can be used to access MPEG-7 based multimedia services from a mobile device. Based on these LAS mobile services, the authors demonstrate the scenarios how user communities might use handheld devices to share MPEG-7 based multimedia services among themselves in order to reduce communication costs or to set up an ad-hoc community network. They propose the enterprise service bus (ESB) technology as an alternative middleware approach to provide multimedia services to mobile devices.

From accessibility point of view of mobile services, Nokia in its research of Mobile Web Server [81] pointed out the addressability issue of mobile devices over operator networks. According to their findings, providing direct IP access to mobile devices is a bottleneck in provisioning M2M service networks. The operators typically employ firewalls to prevent http access to subscribers devices over the Internet. This fact can be observed from [19] explaining how M2M multimedia streaming went through over two big operator networks, O2 and T-Mobile. Table 2.5 depicts the summary of live network transactions. It is observed that in case both client and server devices are connected on private IP addresses, M2M streaming is not possible unless an intermediate access gateway is involved. Furthermore, the direct streaming went successful in case the server is on public IP address with the exception that operator restricts the session explicitly.

### 2.2.5 Quality of Service in Mobile Networks

3GPP Release 5 was the first one to introduce the IP Multimedia Subsystem (IMS). The service providers use the IMS as a standard architecture for a cost-effective

| Client Network | Client IP | Gateway | Server Network | Server IP | Result |
|---|---|---|---|---|---|
| O2 (HSPA) | Private | NA | T-Mobile (HSPA) | Private | Failed |
| T-Mobile (HSPA) | Private | NA | O2 (HSPA) | Private | Failed |
| O2 (HSPA) | Private | Relay | T-Mobile (HSPA) | Private | Sucessfull |
| T-Mobile (HSPA) | Private | Relay | O2 (HSPA) | Private | Successfull |
| O2 (HSPA) | Private | Relay/NA | T-Mobile (HSPA) | Public | Successfull |
| T-Mobile | Private | NA | T-Mobile (HSPA) | Public | Successfull |
| T-Mobile | Private | Relay | T-Mobile (HSPA) | Public | Failed |

TABLE 2.5: M2M streaming results over live networks [19]

network for the growth of IP based communication and media services [46]. Service creation can be quite complex as well as exorbitant expensive for operator and nevertheless for user without IMS. IMS uses SIP as its communication protocol. Moreover, it efficiently reuses available Internet standards. Issues such as QoS, multimedia service charging and control policies are better defined due to the standardization of IMS and supports the operators in this regard. The following sections present the significance of QoS in mobile networks and the forthcoming future infrastructure especially in IMS. Additionally, some schemes available for quality control, QoS differentiation and policy management are also discussed.

The focus of the end user is now on a seamless multimedia content delivery as the wireless market is on the edge of a new era. Consequently, the network providers are really putting their heart to add value to the techniques they are offering to the end user. Moreover, the mobile data usage is increasing exponentially and the 3G/4G supported devices (notebooks, laptops, smart phones etc.) have made competition more interesting for the network operators to comply with the ever increasing demands of the end user. The results of a study conducted by Alcatel-Lucent research in Figure 2.11 shows how the user behavior reflects its increasing dependency on wireless data. The figure shows the monthly web browsing statistics of smart phones up to 70MB, super phones up to 100 MB and the tablets up to 350 MB. [64]

This statistics unveil a challenge to the consumer market for improved portable devices production, to network operators for improving network experience and application developers for producing better applications leading to good QoE.

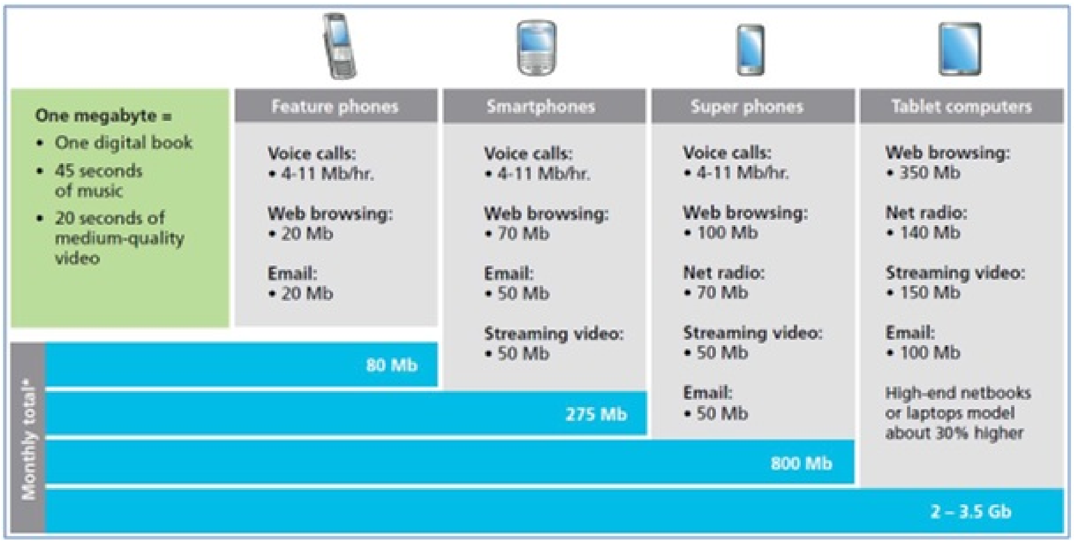



FIGURE 2.11: User behaviour and wireless data relationship per month [64]

#### 2.2.5.1 QoS in Cellular Networks

End-to-end QoS is considered between Terminal Equipment (TE) by example of UMTS Bearer services are applied on various protocol layers for providing users with better network quality. A bearer service with suitable functionalities needs to be set up for non-destructive communication between source and destination. Two bearer services are mentioned below [23]:

The provision for signal and user data transportation between Mobile Terminal (MT) and the Core Network (CN) is ensured by Radio Access Bearer Service. It provides error protection.

UMTS CN nodes are connected to the CN Gateway for communication to external nodes via Core Network Bearer Service. Also, the backbone network (IP network consisting of GGSN and SGSN nodes) is controlled and utilized by the Core Network Bearer service. This is done quite efficiently and effectively to provide contracted UMTS bearer service based on a set of four different QoS classes:

- Conversational Class deals with extremely delay sensitive real time traffic e.g. voice and video telephony.
- Video streaming belongs to the Streaming Class which is highly sensitive to delay and needs reliable network resources.

- Web Surfing, Email etc. lie in the Interactive Class which is pretty less sensitive as compared to conversational and streaming classes.
- Background Class is known as the most delay insensitive class and is mostly used by the applications running in the background.

QoS in fact is an important issue in 3GPP standardization, see [39].

#### 2.2.5.2 QoS technical Key Performance Indicators (KPI)

Following KPIs are helpful for the Technical QoS measurements by service providers. [37]

- **Delay** Network delay is a major factor in user satisfaction. Once a network session is established, the delay is the time taken by the network to successfully deliver the content from source to destination. In case the network is burdened, data packets in long queues of IP networks experience considerable delay.
- **Jitter** The variations in network delay is referred to as Jitter. It has a great influence on the end user perception of a delay sensitive service.
- **Throughput** Bits or Bytes per second are the unit for throughput in a network. QoS greatly depends on the throughput received by a user device over the network. Most of the time, it turns out quite hard to prioritize resources for particular applications like multimedia services. It is because of the ups and downs in the traffic load over the shared medium. Consequently, the user experience is relegated in the congestion time.
- **Bandwidth** Bandwidth of radio channels is shared between the users and is also considered a key parameter of QoS. It tells about both the available and busy resources for traffic in a network. The concept can better be understood by the example of a user who is trying to access a videophone paid service within a WLAN locale. He would require a reliable bandwidth for uninterrupted usage; otherwise, he would be truly disappointed and frustrated.
- **Packet Loss** Packet loss rate is inversely proportional to user satisfaction. It is an important factor of QoS. Packet loss in IP network occurs mainly during the routing process between the service provider server and the end user or it

may also occur due to some hardware problem. Both generally result in the loss of information. The packets are retransmitted in case of packet switch network, increasing the network delay even more. For example when some VoIP or video content is requested over the wireless network and it faces packet loss, the user perception may be considerably reduced.

- **Bit Error Rate (BER)** Some environmental factors like noise, fading and interference also have a bad influence on the wireless networks and cause receiver degradation due to errors in transmitted bits.

#### 2.2.5.3 QoS measurement technique

iQoS - Individual QoS [23] is a calculation technique in cellular networks which calculates per user satisfaction rate. This per user satisfaction rate is a vital parameter. The iQoS scheme prioritizes the users on the basis of bad QoS so that the users facing issues can be pointed out and accommodated for corrective measures at earliest. This whole process is followed in order to provide the users with improved services.

Users who are in close vicinity of a base station have better radio signal quality than those who are more distant. Users who face poor signal quality may suffer from call dropping and call blocking. Their QoS can be improved by iQoS measurement and proper reaction by the mobile network.

#### 2.2.5.4 Session based QoS

The IMS assists the network and service providers to deal with user/session based policies. In this section, we illustrate three variants of QoS support: Connection Based QoS, Flow Based QoS and Session Based QoS. [34].

- **Connection Based QoS** The connection based quality mechanism is inept for the input and output, scalability and the overall performance. This type of QoS necessitates absolute bandwidth (a circuit switched channel) before any data transaction is completed. [34]

- **Flow Based QoS** It supports the relative priority policy of packet switched data flows and cannot be controlled without the help of the edge devices.

Though, both congestion control as well as low precision are a dare in flow based QoS. [34]

- **Session Based QoS** The IMS network offers the session based QoS. The Core Server generally controls the whole transaction. Quality parameters attend to per user, per session and per service basis. Differentiated policy supports in resource optimization and it makes the best network resource usage worthwhile. [34]

#### 2.2.5.5 E2E QoS Function Distribution

The IMS embraces the data path from UE to the core network. It is also known as the open standard for End-to-End QoS. IMS, LTE UE, RAN, EPC and IP Network are the five sections in E2E QoS Functions which additionally can be linked with the open Internet. It is shown in the figure 2.12, whereas, table 2.6 illustrates the important QoS functions of these sections.

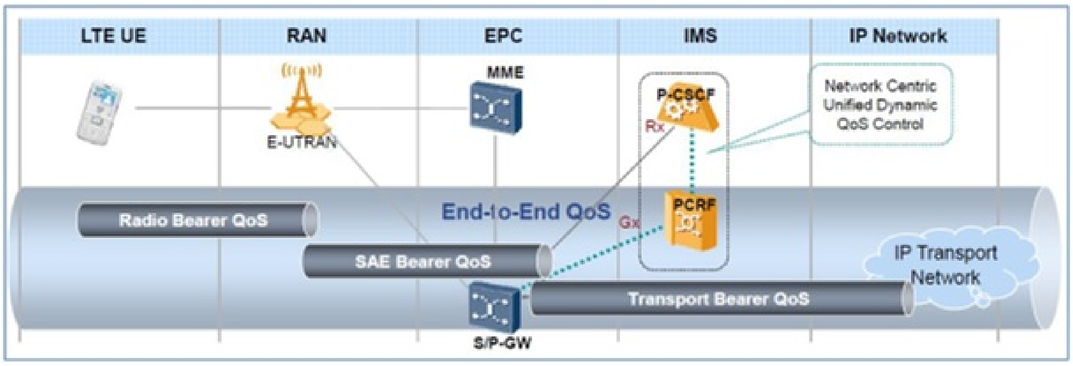



FIGURE 2.12: E2E QoS Function Distribution [34]

#### 2.2.5.6 QoS differentiation concept by Nokia Siemens Networks

A study was performed by Nokia Siemens Networks (NSN) [37] for the growing need of QoS solution and it presented its viewpoint of QoS differentiation. An operator can dispense the bandwidth in the network via over dimensioning which is believed to be a practicable QoS solution. QoS differentiation adds to performance monitoring by analyzing segregated flow of traffic on peculiar basis rather than monitoring single aggregated flow. QoS differentiation leads to QoS profiles in a

| | |
|---|---|
| LTE UE QoS Function | Visual Quality of Experience (VQE) functions handling, e.g. jitter buffer, etc.<br>Request initiation of Service QoS<br>Radio optimization technology for VoIP applications |
| RAN QoS Function | Perform radio optimization for VoIP applications Radio resource management including: packet allocation, scheduling, filtering, etc. |
| EPC QoS Function | Handle the VQE functions<br>Perform the service bearer setup for EPS<br>Accomplish the traffic shaping and IP filtering<br>Enable the PCEF functions<br>Monitor the mapping of QoS |
| IMS QoS Function | Enabling the PCC QoS<br>Retrieve subscriber QoS profile from repository for policy implementation |
| IP Network QoS Function | Enforce the QoS based on the subscriber profile<br>IP transport QoS control along with guaranteed resources to achieve better user experience |

Table 2.6: E2E QoS Functions [34]

network however, the total number of QoS profiles depends on the requirement of the operator. Following are the advantages that most of the operators get:

- High (or Sufficient) QoS
- Emergency Services
- More efficient use of resources
- Monitoring traffic streams
- Flexible support of service offers and business strategies

In case of mobile network capacity e.g. air interfaces are shared among active users within an area and a capacity upgrade involves high charges. So, in this type of cases, the solution that is considered the most favorable one is QoS differentiation. Figure 2.13 shows the three proposed components of QoS differentiation: QoS Management, QoS Control and QoS Enforcement. QoS profile specific monitoring in the system is dealt with the help of QoS Management. For visualization and additional network handling, the management tools get the QoS profile information from the network entities.

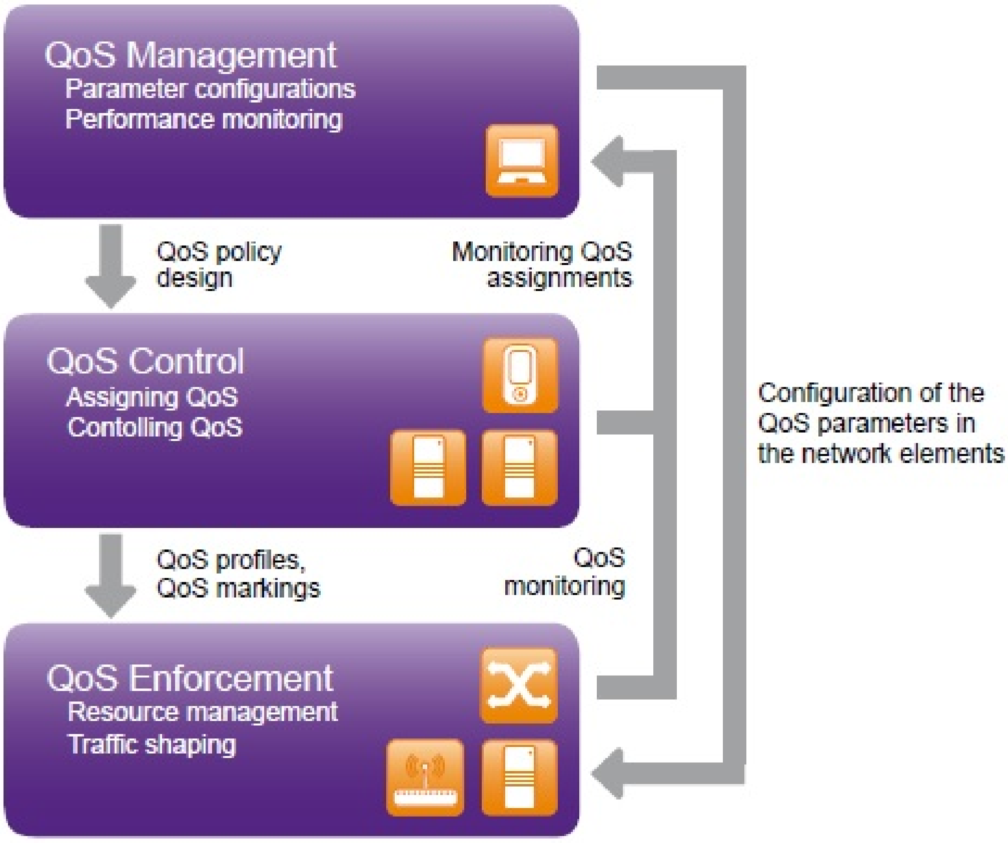



FIGURE 2.13: NSN - QoS differentiation solution and their relationships [37]

QoS Control is responsible for the mapping of an application or a subscriber with a particular QoS. This component also deals in preventing an application for avoiding manhandling of the network QoS.

Efficient allocation of the network resources is the part that is performed by QoS Enforcement. QoS controller sets a QoS which is followed by the network elements in a cellular network however they are not aware of the contents of the data.

## 2.3 Conclusion

This chapter briefs the history of cellular network and explains how technology evolved from the basic telephony (circuit switching) to packet switching and all-IP networks. It is worthy to note that the concept of mobile packet data services was first time presented in 1985 [77] after the introduction of basic telephony in 1876.

This means the advanced data networks (all-IP networks) took only about 30 years to reach at this stage.

The state of the art section presents example of advances made by the industry (Ericsson, Huawei, Alcatel Lucent) and research (e.g. Fraunhofer Fokus) in the IMS and LTE mobile networks domain. It has been identified that there exists no much work explicitly presenting the concept and architecture of mobile application server in the operator LTE/IMS network. The quality of service concept, parameters and issues in mobile networks are extensively studied and some strategies are identified to tackle these issues. Based on this conclusive study, the dissertation work has utilized the following state of the art tools and concepts in the realization of M2M service network :

- The base architecture of Mobile Server Platform [9] has been extended for standardized cellular interfaces and protocols of LTE-EPC system.
- Ericsson testbed for IMS ecosystem [60]
- Ericsson test agents for SIP and VoIP services [60]
- Fraunhofer testbed for EPC ecosystem [49]
- Session based QoS as a QoS measurement technique [34]
- QoS differentiation solution similar to proposed by Nokia Siemens Networks [37]
- Policy and charging framework devised by 3GPP [38]

# Chapter 3

# IMS - Mobile Server Platform

This chapter enfolds the novel concept of Mobile Application Servers as a foundation of M2M service networks in the cellular world. The chapter starts with a discussion of what are the potential challenges faced while realizing the M2M service networks over cellular operator networks. Then, a complete solution based on the existing IMS and LTE- EPC mobile networks is proposed. The work presented in this chapter has first been published by the author in [19].

## 3.1 Challenges in M2M Service Networks

M2M service provisioning in cellular networks is not trivial; the proposal has to integrate numerous communication standards and protocols, address organizational interests and benefits, and to assure the quality of service requirements for end users' satisfaction at the same time. With the help of an example scenario depicted in Figure 3.1, this section explains the three major challenges that one has to face in deploying M2M service networks over the cellular operator network. The two mobile subscribers A and B (acting as service clients) are trying to access web services hosted by another mobile subscriber (C, acting as a mobile web server) over an operator network (T-Mobile).

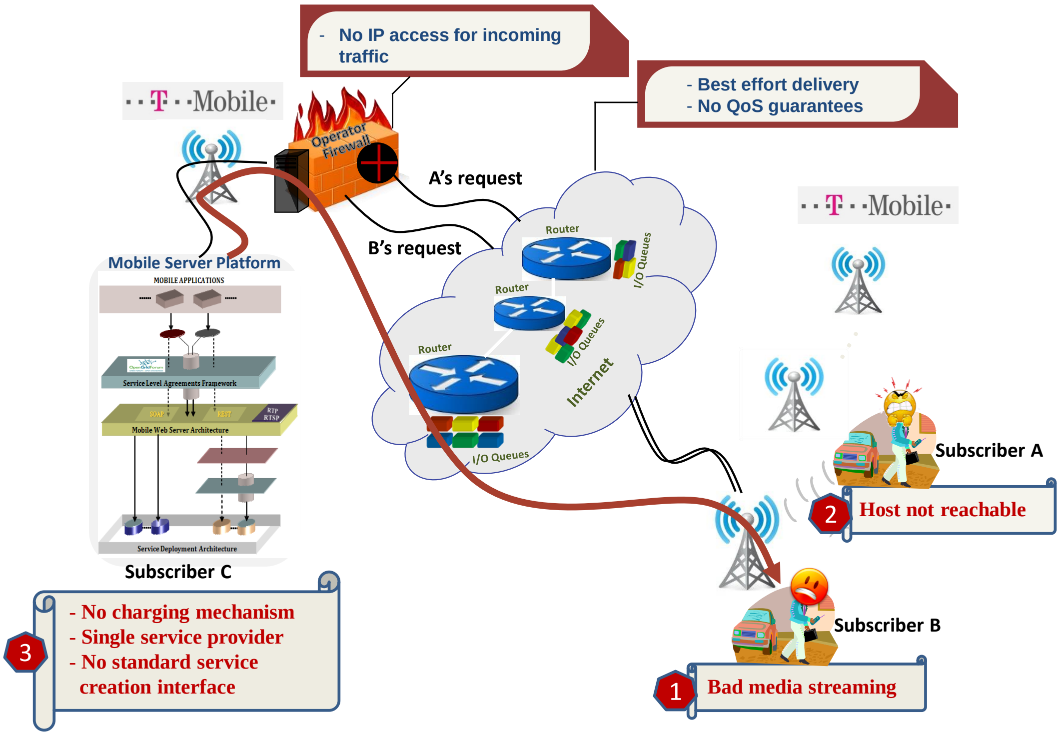



FIGURE 3.1: Issues in M2M service networks

### 3.1.1 Quality of Service Issues

Subscriber A wants to access a multimedia streaming service hosted by subscriber C. Assuming no operator firewall between A and C exists, the service request from A reaches C through the operator data network without any problem. In response to the request, subscriber C initiates its streaming service, creates a multimedia session with A and starts sending multimedia data towards A. Due to congestion in the network, the data gets delayed (badly) as well as some packet loss occurs before it reaches subscriber A. Hence, subscriber A experiences a poor quality multimedia streaming at his end.

The scenario above is an example of cellular operators' lack of control and management over the services running in their network. Without identifying the type of data, they apply best-effort service delivery policy on their network i.e., the network provides no guarantees of seamless and in-time delivery of data to its destination. Best-effort policy is an option also in postal service, where the in-time delivery of a certain letter is not promised. Under extreme load when too many letters arrive to the post office, the letters get delayed due to lack of resources.

On one hand, the best-effort policy is a good option to disseminate text data services across the network such as Email service and text messaging service. These services are categorized as delay tolerant, where in-time delivery of data is not so important. Even some data loss is bearable by these services to some extent given the context of the text is not violated. On the other hand, a best-effort network does not seem to comply with the requirements of multimedia service deliverance. Unlike textual services, multimedia services are known to be bandwidth hungry and delay in-tolerant by nature. Therefore, distribution of multimedia content over cellular networks requires some special treatment i.e., the underlying network must guarantee a certain level of constant data speed for a given multimedia session. Network over-provisioning to achieve this is the usual solution but not an economical one in this regard.

### 3.1.2 M2M Service Blocking

In figure 3.1, subscriber B also wants to access a service hosted by subscriber C. Subscriber B sends a service invocation request towards C and waits for a response. The request goes through the network but gets blocked by the operator firewall. Eventually,subscriber B receives a “host not reachable” message by the network.

This special case is an example of operational level issue in M2M service invocation, where a subscriber device is not permitted to receive an incoming HTTP request from inside and outside the network. This means the cellular operators do not encourage the subscriber devices to open server sockets at their network in order to become a service hosting machine. Hence, the operator firewall blocks every incoming session to subscribers’ device which is originally invoked by some other device inside and or outside the network.

The most probable reason behind this operator policy is to restrict the OTT services, as discussed in section 1.1. As mentioned earlier, a web server machine is expected to receive hundreds of requests at a time from inside and outside the network. Therefore, an operator does not permit any subscriber device (acting as a server) to overburden the network by increasing the data traffic and over-utilizing network resources in order to serve the incoming service requests. Hence, considering the cheap flat-rate data tariffs of cellular operators, how can an operator tolerate an individual subscriber to become an OTT-type server and over-utilize resources.

There are some state-of-the-art techniques useful to bypass NATs (Network Address Translation) and network firewalls. Such techniques are e.g. applied by OTT service providers to bypass operator firewall restrictions in cellular networks. However, as discussed in section 1.1, operators see this OTT model as a threat to their business ecosystem unless they get some real benefit out of it.

### 3.1.3 M2M Service Platform

The M2M conceptual model is somewhat different from a typical client-server model, where a server machine in the network is dedicated for serving clients connected to the network. In an M2M service provisioning model each node connected to the network is aimed to have the ability to become a service host (server) or a service client at the same time. Hence, they require a Mobile Server Platform (MSP) that has the ability to act both as a server and a client simultaneously.

This case study sees it as a big challenge to realize an MSP that conforms to the standards of the cellular industry. Therefore, it identifies some main limitations of the MSP introduced in [9] to be used as a standardized service delivery platform in the foundation of M2M service networks for cellular systems:

- The MSP platform does not specify a standardized way to register itself as a client or an application server with the cellular data network.
- It does not define the service discovery and service invocation mechanisms by using the standardized interfaces and protocols of the cellular system.
- It does not integrate with the standardized policy and charging solution of an operator data network.

## 3.2 Realization of M2M Service Networks

In order to address the aforementioned challenges, this section presents the core concept of the dissertation that lays the foundation of M2M service network in the LTE mobile system. The concept is to utilize in mobile nodes the concept of IP Multimedia Subsystem (IMS), specified for the application layer of LTE systems, which originally was designed to enable cellular operators to host and control their own applications in the network, and apply their own QoS and charging policies accordingly.

### 3.2.1 IMS - Mobile Server Platform

As explained earlier in section 3.1.3, an MSP is a part and parcel of an M2M service network. It is a main source of sharing data and services among network nodes. Hence, from a cellular network perspective, it has to adopt and integrate with many standardized interfaces and protocols in order to become both a service host and service client in the network.

This section explains the design and architecture of an IMS based Mobile Server Platform (MSP), which lays the foundation of M2M service networks in the LTE mobile system. As shown in Figure 3.3, the proposed MSP is comprised of two main modules: The IMS - Mobile Application Client (MobAC) module and the IMS - Mobile Application Server (MobAS) module. As explained earlier in previous chapter (section 2.2.2), Application Servers (ASs) are the entities responsible for hosting and executing multimedia services in the IMS network, which can be of three different types [62]: SIP AS, Open Service Architecture (OSA) AS and CAMEL AS. Although some basic requirements are identified, 3GPP does not specify the underlying architecture of these servers. However, some software designs have been presented by research communities based on the state-of-the-art APIs and technologies targeted mainly for enterprise solutions such as [61]. Similar to SIP ASs in the IMS network, the dissertation introduces a concept of MobAS (can also be referred to as MobSIPAS), which is a light-weight SIP application server designed for mobile devices enabling them to host and execute M2M services in the LTE packet core, LTE - EPC.

The basic design of the MSP is a modified version of the REST - interface Mobile Web Server framework proposed by Aijaz [9, 10, 12]. The modifications are mainly introduced in order to incorporate the IMS signaling and request-response mechanisms proposed by 3GPP. For instance, SIP is used for signaling in the IMS. Therefore, a SIP server interface is a natural choice for the MSP to receive service invocation requests coming through the IMS network. However, the introduction of SIP server interface does not let the base design of the MSP to be purely Restful (as in [9]) i.e., HTTP based service requests are not directly supported. Hence, SIP bindings are created with REST URIs as explained in the following subsection.

Similarly, SIP MESSAGE and SIP INVITE are two basic request types that are used to initiate instant messaging and voice/video calling service in the IMS. Thus, mapping of such SIP methods is required in the MSP to the underlying synchronous (for short-lived communication) and asynchronous (for session-oriented long-lived communication) service invocation mechanisms of RESTful architecture (as in [9]). Hence, SIP MESSAGE request is mapped to synchronous service invocation mechanism for its session-less short-term nature, whereas SIP INVITE request is mapped to asynchronous service invocation mechanism for its session-oriented nature. However, as the asynchronous service invocation mechanism operates on a request-response and solicit-response principle (as depicted in 3.2), the three way handshaking mechanism of SIP INVITE request cannot be fully supported without extending the basic asynchronous design. This phenomena is further explained in section 4.2.2.

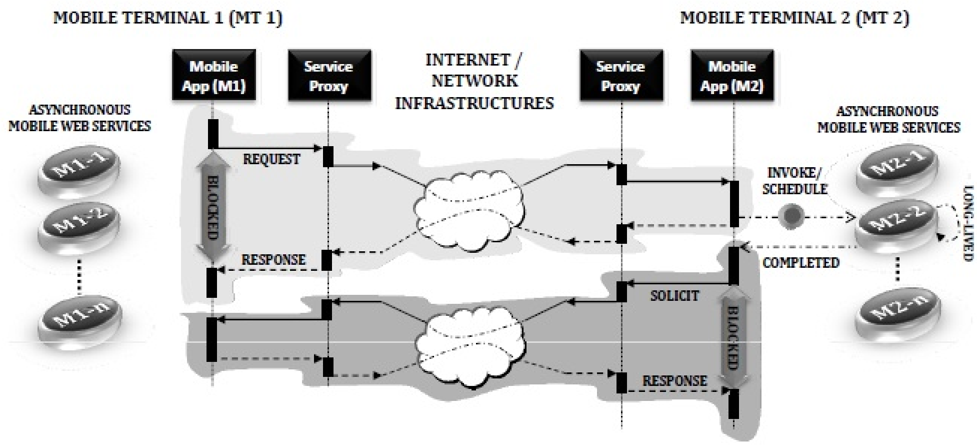



FIGURE 3.2: An M2M scenario depicting the Request-Response and Solicit-Response operations [9]

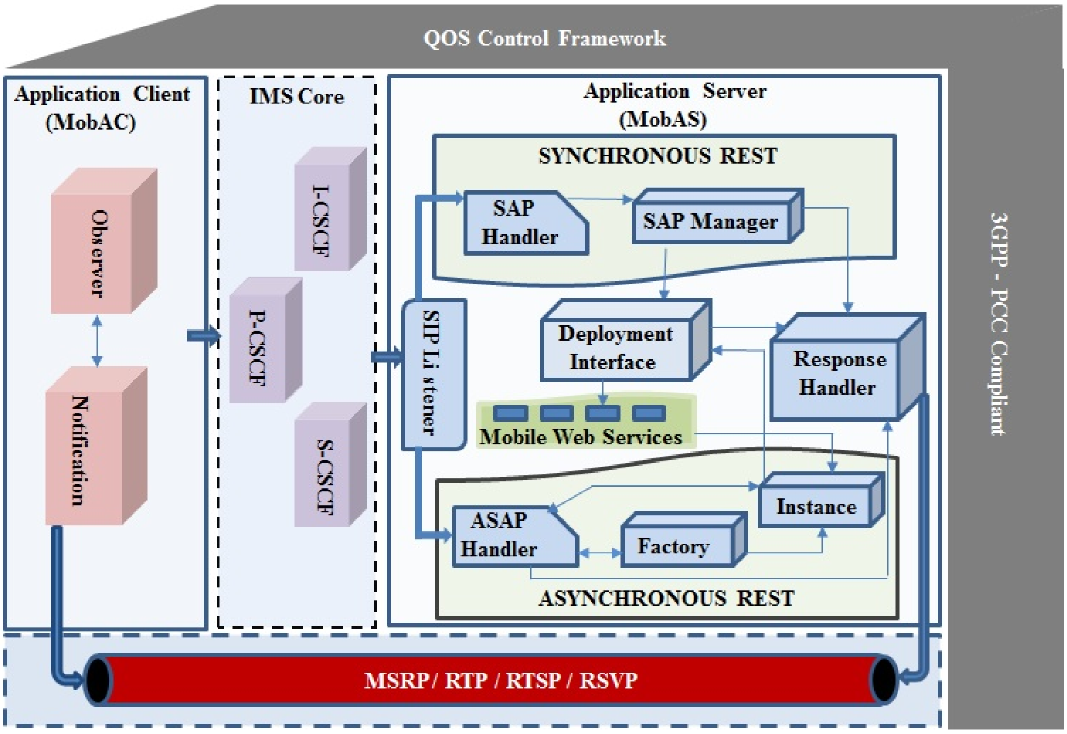



FIGURE 3.3: Mobile Server Platform located on mobile device

#### 3.2.1.1 SIP Requests Binding with REST URI

The first and the foremost requirement of integrating the MSP with the IMS ecosystem is the implementation of SIP protocol. SIP (Session Initiation Protocol) is used to create multimedia sessions in most of the multimedia streaming applications of Internet world, such as Voice over IP (VoIP) and IP conferencing applications. Therefore, in order to ease the integration with most of the Internet applications, SIP was adopted by 3GPP to support multimedia applications in the IMS.

However, the introduction of SIP in MSP is not straight forward. In order to cooperate with the underlying REST style service creation and invocation process, a SIP request has to incorporate a RESTful URI which is a major source of locating a particular resource (restful mobile web services) in the service deployment interface. Hence, the dissertation proposes a new header field, named as rest-uri, in the basic SIP MESSAGE and SIP INVITE messaging constructs.

Two types of SIP requests based on synchronous and asynchronous system are discussed below. For more details on REST URI format for synchronous and asynchronous services, see [11].

**Synchronous SIP Requests** The SIP request methods used to create synchronous service calls are termed as Synchronous SIP requests. For example, SIP MESSAGE and SIP Bye requests are used to invoke s1 and s2 synchronous services in Figure 3.5. Listing 3.1 shows a synchronous SIP request where a SIP user Alice wants to send a text message to another SIP user Bob. It is noticeable that the request contains a 'rest-uri' header with Restful URI pointing to a synchronous service for Instant Messaging, IMSync.

```
MESSAGE sip:greetings@bob-server.com SIP/2.0
Max-Forwards: 69
CSeq: 1 MESSAGE
resourcemethod: GET
P-Called-Party-ID: <sip:greetings@bob-server.com>
Content-Length: 84
Call-Info: <http://www.antd.nist.gov>
rest-uri: POST/IMsync
Contact: "alice" <sip:alice@10.0.2.15:5070>
Record-Route: <sip:127.0.0.1:5082;from-tag=12345;lr>
P-Asserted-Identity: <sip:alice@ericsson.com>
To: "bob" <sip:bob@ericsson.com>
From: "alice" <sip:alice@ericsson.com>;tag=12345
Call-ID: 79a6e89a0f8c55849222730b2a90769f@10.0.2.15
Via: SIP/2.0/udp 127.0.0.1:5082;branch=
z9hG4bKaba7c1f00b8133b045de689d1ca1b46e,
SIP/2.0/UDP 10.0.2.15:5070;branch=
z9hG4bK29cb37aa1e68dd24e29b2ca0762fae9a383639;
received_port_ext=5081;received=192.168.1.4
Content-Type: text/xml
<?xml version='1.0' ?><RESTRq><Rq><demo>Hi Bob,
alice here!</demo></Rq>
</RESTRq>
```

Listing 3.1: Synchronous SIP Request

**Asynchronous SIP Requests** The SIP request methods used to create asynchronous service calls are termed as Asynchronous SIP requests. For example, SIP INVITE request is used to invoke an asynchronous service s3 in Figure 3.5. Listing 3.2 shows an asynchronous SIP request where a SIP user Alice wants to send a multimedia file to another SIP user Bob. Here the 'rest-uri' header points to an asynchronous service for multimedia Instant Messaging, IMasync.

```
INVITE sip:greetings@bob-server.com SIP/2.0
Max-Forwards: 69
CSeq: 1 INVITE
resourcemethod: GET
P-Called-Party-ID: <sip:greetings@bob-server.com>
Content-Length: 751
Call-Info: <http://www.antd.nist.gov>
rest-uri: POST/Request-Response/Factory/
createInstanceRq/IMasync/add
Contact: "alice" <sip:alice@10.0.2.15:5071>
Record-Route: <sip:127.0.0.1:5082;from-tag=12345;lr>
P-Asserted-Identity: <sip:alice@ericsson.com>
To: "bob" <sip:bob@ericsson.com>
From: "alice" <sip:alice@ericsson.com>;tag=12345
my-other-header: my new header value
Call-ID: 37c08f9666a687a3027ac22b98483158@10.0.2.15
Content-Type: application/sdp
Via: SIP/2.0/udp 127.0.0.1:5082;branch=
z9hG4bK4a2b443a5792a53b7d13c83662dd87cf,
SIP/2.0/UDP 10.0.2.15:5071;branch=
z9hG4bK3054372ee7534b8d107a936cd29931e1393030;
received_port_ext=5081;received=192.168.1.4
v=0
o=coco 1326930823911 1326930823911 IN IP4
bob@ericsson.com
s=sdp offer (recvonly)
c=IN IP4 sharique@ericsson.com
t=0 0
m=message 9097 TCP/MSRP
a=recvonly:recvonly
a=accept-types:message/cpim
a=accept-wrapped-types:*
a=path:msrp://bob@ericsson.com:9097/jshA7we;tcp
a=file-name:multimedia.jpg
a=file-transfer-id:aCQYuBRVoUPGVsFZkCK98vzcX2FXDIk2
a=file-size: 38503
```

Listing 3.2: Asynchronous SIP Request

#### 3.2.1.2 Mobile Application Client

As shown in Figure 3.3, Observer is an integral component of MobAC which is responsible to create and send synchronous and asynchronous service calls. The synchronous service calls are exposed based on the request-response interaction policy of WSDL operations [80]. In a request-response process each client request is followed by a response. Therefore, the Observer keeps itself in a blocked state after a synchronous service call until the response is back from the requested service invoked by the MobAS. On the contrary, the mechanism of asynchronous service calls is derived by incorporating the request-response and solicit-response WSDL operations [80]. Similar to request-response operation, solicit-response is a process where every request is followed by a response, but the roles of service provider and service consumer are reversed.

In contrast to synchronous calls, the Observer does not wait for a complete service response after an asynchronous service call, rather an acknowledgment is sent by the MobAS to the Observer that the request is received. However, during this request-response operation the Observer provides an End Point Reference (EPR) to the MobAS where the service outcome shall be notified. After this operation, the service starts running independently in a separate thread sending response back to the provided EPR in the form of short notifications with partial results time to time or a single complete response at the end of successful task completion. This process of result notification in asynchronous service calls is analogous to the solicit-response operation. Typically, an asynchronous interaction is utilized to realize the session based services such as multimedia messaging and streaming, where the initial request-response operation is used to create session with an additional acknowledgment message and the solicit-response approach is used for actual media delivery by the service.

#### 3.2.1.3 Mobile Application Server

In Figure 3.3, the MobAS module starts working when it receives a SIP request from a MobAC via the IMS network, which is the SIP signaling network in between MobACs and MobASs that are connected by the IP-based operator network that provides PCRF. The SIP Listener separates the Restful-URI string from the SIP-based service request and parses it to extract the target service information such as service name, request method and resource method. The parsed information is then utilized by SIP Listener to form a Request Object and identify the service type such as synchronous or asynchronous. It is worth mentioning here that although the service type can alse be identified by simply knowing the SIP method name, such as, SIP MESSAGE corresponds to synchronous request and SIP INVITE corresponds to asynchronous, this approach is avoided in order to keep compliance with the basic REST architecture introduced in [11].

#### 3.2.1.4 Synchronous Service Invocation

In case the synchronous service request identified by SIP Listener, the Request Object is passed to the Deployment Interface through SAP (Synchronous Application Processing) Manager which is responsible for look up and invocation of the target service. The Deployment Interface maintains a list of all available services along with their corresponding objects as a key-value mapping. Therefore, it looks up the corresponding service object in its service inventory and use it to invoke target service.

#### 3.2.1.5 Asynchronous Service Invocation

An asynchronous service is initiated and executed as a thread, which is the primary difference between the synchronous and asynchronous services. The asynchronous service is made a thread because that enables it to be controlled and monitored and to be in variable states during the course of execution. In case of asynchronous service request, before parsing and making Request Object the SIP Listener first checks whether the request message contains an SDP (Session Description Protocol) offer, extracts the media information and verifies the availability of the target service. An SDP offer is used in the SIP INVITE request in order to negotiate over streaming media parameters between two parties before the actual media transmission starts. The SDP works in conjunction with RTP and MSRP protocols for real-time streaming and multimedia instant messaging services. The SDP headers and messaging constructs are defined in the IETF specification [54].

Considering the SDP offer is valid and accepted by the SIP Listener, the Request Object is passed to ASAP (Asynchronous Application Processing) Handler which invokes Service Factory to create a new Service Instance for the target service. At this stage, a 200 OK message is sent to the AC that releases the AC from the blocked state. With this message the AS also provides two important IDs to the AS: a request Call-ID for reference to the upcoming final acknowledgment from the AC and the EPR for newly created service instance for future communication with the service.

Thus, after sending 200 OK message the control is given to the Deployment Interface finally along with the Request Object and Service Instance parameters for target service invocation. In contrast to synchronous service invocation, after inventory look up the Deployment Interface forms a service thread, save it in database against the Call-ID and wait for the acknowledgment of the last 200 Ok message from the AC. Consequently, when the acknowledgment message arrives at Request Handler, the Deployment Interface is notified along with the Call-ID which in return starts the corresponding service thread.

### 3.2.2 M2M Service Provisioning

This section describes how the IMS architecture helps the subscribers' devices to make one-to-one contact over the operators cellular network. The dissertation proposes that M2M services are to be invoked in a client server fashion where one ( client) of the two communicating parties makes request and the other party (server) serves the request and invokes the requested service.

For every short or long process to be performed by the mobile server, there exists a service categorized as synchronous(short-term) or asynchronous(long-term) respectively. Therefore, by defining a proper mechanism how MobASs and MobACs may register themselves in the operator's network, this section enfolds how an IMS compliant MSP can be used to realize an M2M chat application over the cellular operator network.

#### 3.2.2.1 Mobile Application Server Registration

Figure 3.4 glimpses an Ericsson SDS (Service Development Studio) interface, which is an Ericsson's testbed to design, create and test IMS services by the network operators and independent software vendors. The figure shows how ASs are registered in the DNS system of the operator network. Every AS is assigned a unique SIP URI which is translated to a static IP address and a port number. For instance, an Ericsson's Presence and Group Management (PGM) AS is assigned a URI as "sip:pgm.ericsson.com", IP address as "127.0.0.1" and port number as "5060". Here the IP address is pointing to the local server/machine.

Hence, the dissertation proposes this same mechanism to register subscribers' devices as the MobASs in the IMS network. Thus, an operator may assign the static IPs to the subscribers' devices who wish to introduce their devices as MobAS in the network. An alternative and more flexible approach would be to assign the static IPs to the user profiles instead of particular devices. So, whenever a certain user is logged into the operator's data network, the network assigns the same IP address every time. However, these are the policy related matters which are dependent to the network operators to what they decide for their network.

#### 3.2.2.2 Mobile Application Client Registration

The client/user registration in the IMS network is the same as the standardized registration process used by any SIP application i.e., to send a SIP REGISTER request to the network and wait for a 200 OK response back. Thus, in perspective of MobAC, this is a responsibility of the Observer (discussed in section 3.2.1.2) module to send a SIP REGISTER request to the IMS network (P-CSCF), whenever a user logs in to the MSP.

Listing 6.1 shows a sample SIP REGISTER message to be sent to the IMS network. It includes all the key headers and their corresponding values which are used by the IMS-CSCF servers in order to understand the request. However, the authorization and QoS headers are not shown in this listing that will be covered in chapter 6 in detail. Below mentioned is a quick overview of some key headers of the SIP REGISTER request:

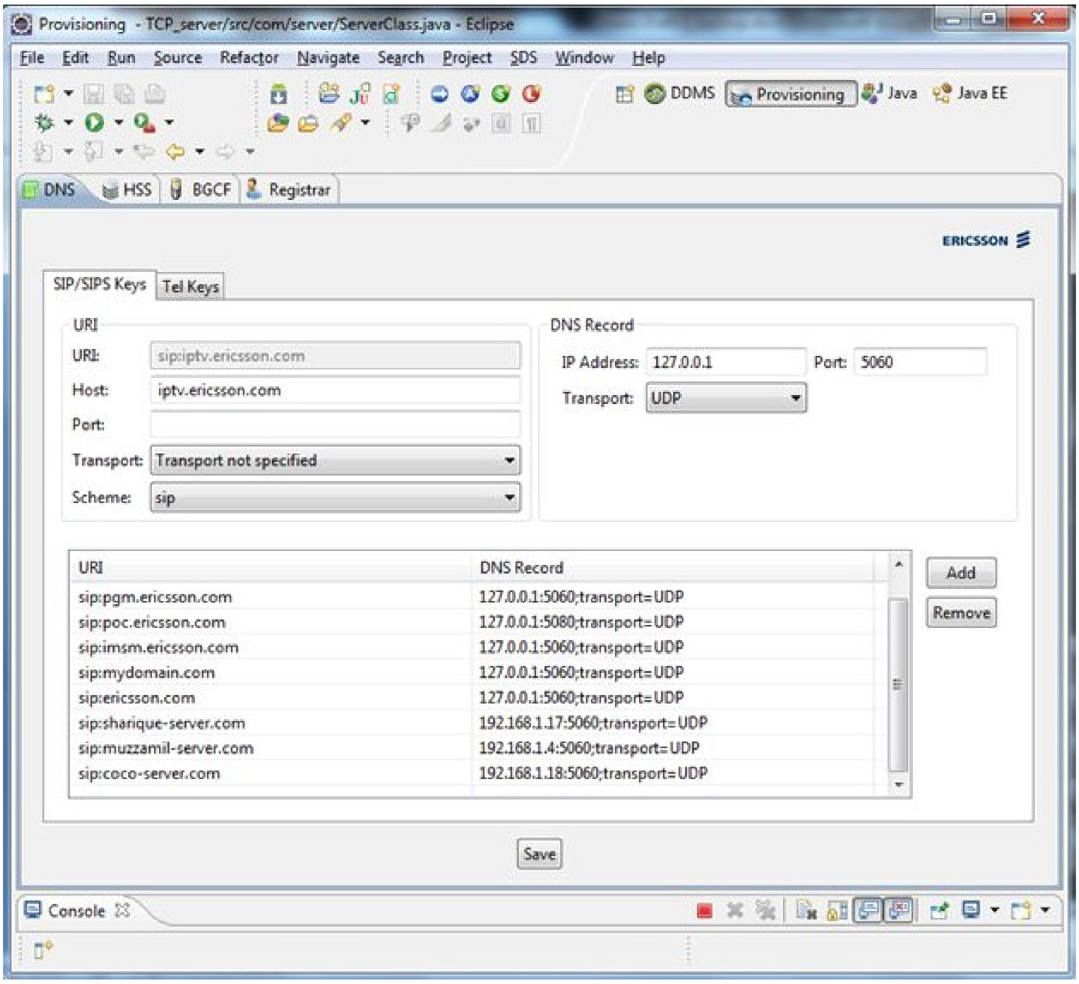



FIGURE 3.4: Mobile Application Server Registration to the Ericsson IMS testbed

***Request Line*** indicates the type of SIP requested method and request uri which is to be sent to the IMS framework. In the given example the register request indicates : **Method**= REGISTER, **Request-URI**=IMS Framework domain (e.g. sip:open-ims.test)

***Via:*** It contains the local address of user (bob) i.e., domain (e.g. 192.168.2.105:6060) where it is expecting the responses to come.

***Max-Forward:*** It is used to limit the number of hops that this request may take before reaching the recipient. It is decreased by one at each hop. The example in listing 6.1 shows a count of 70.

***To:*** It contains a user name (”bob”) and a SIP URI < bob@open-ims.test >

***From:*** It also contains a user name "bob" and a SIP URI < bob@open-ims.test >. It also contains a tag (e.g. tag=12345) which is a pseudo-random sequence inserted by the SIP application. It works as an identifier of the caller in the dialog.

***Call-ID:*** It is a globally unique identifier of the call generated as the combination of the IP address and a pseudo-random string. The Call-ID is unique for a call; however, a call may contain several dialogs. Each dialog is uniquely identified by a combination of From, To and Call-ID.

***CSeq:*** It contains an integer and a method name. When a transaction starts, the first message is given a random CSeq. After that it is incremented by one with each new message. The CSeq header is very useful and it is used to detect non-delivery of a message or any message which is outdated or out-of-order.

***Contact:*** It contains a SIP URI that is a direct route to user. It contains a username and a fully qualified domain name (FQDN). It may also have an IP address. Moreover, the via field in the SIP message header is used to send the response to the request, however, the Contact field is used to send future requests. That is why the 200 OK response from receiver goes to sender through proxies. But when receiver generates a BYE request (a new request and not a response to INVITE), it goes directly to sender bypassing the proxies.

***Content-Type:*** It contains a description of the message body or an SDP message.

***Content-Length:*** indicates an octet (byte) count of the message body.

```
REGISTER sip:bob@open-ims.test SIP/2.0
Call-ID: fa60d272af3e8d7ef33af7445c99322b@192.168.2.101
CSeq: 1 REGISTER
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "bob" sip:bob@open-ims.test
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bKb6f36916c7b7c863e069f72d6326905a323535
Max-Forwards: 70
Contact: "bob" sip:bob@192.168.2.101:5070
My-Header: my header value
My-Other-Header: my new header value
Call-Info: http://www.antd.nist.gov
User-Agent: UCT IMS Client
Expires: 600000
Supported: path,gruu
```

```
Content-Length: 0
```

Listing 3.3: Initial SIP Register Request

Similarly, for deregistering an already registered user from IMS framework, we need to send again a REGISTER message to the framework. Therefore, we need two major improvements in the SIP header, one is the increment of the CSeq number header and the second is the Expires header field value set to 0. The server analyzes these two header field values and then the CSCF(s) modules will forward the request to the HSS and the user status becomes deregister and a notification message is send back to user.

#### 3.2.2.3 M2M Service Invocation

A two way communication is shown in Figure 3.5 where both devices are capable of becoming service client and server at the same time by using mobile server platform (MSP). The MSP is comprised of the IMS mobile application client (MobAC) and the IMS mobile application server (MobAS). Like SIP ASs in the IMS network, the MobAS is a mobile based SIP application server capable of hosting and executing multimedia services. Section 3.2.1 describes the design of MSP in detail. The figure depicts a chat messaging session between two mobile applications, A1 and A2. The application A1 requests for an image file from A2. The application server AS2 receives a SIP request message from the client AC1, invokes its messaging service and sends a 200 Ok response back to AC1. The application A2 reads the message and picks an image file to be sent to A1. Accordingly, the application client AC2 gets an acceptance from application A1 by sending a SIP request message and receiving response from the server AS1 in the same fashion as above. Until this step (no. 10), the messaging service discussed is a fine example of synchronous mobile service which implements a simple short-lived process i.e., receiving the client request and sending response back immediately without keeping the client in a blocked state for long.

Having the initial negotiation done, the application client AC1 sends a SIP INVITE request to the server AS2 in order to invoke actual media sharing. The AS2 sends a 200 OK response immediately and invokes a multimedia messaging service later on after having an ACK message from AC1. At this point, a direct connection is setup between the service and and the client AC1 and packets start forwarding from service to AC1. Finally, the service closes the connection with a SIP Bye request after the final packet is sent. The multimedia messaging service in this scenario is an example of asynchronous mobile service because the process of sharing multimedia files is somewhat a lengthy process depending on the media file size.

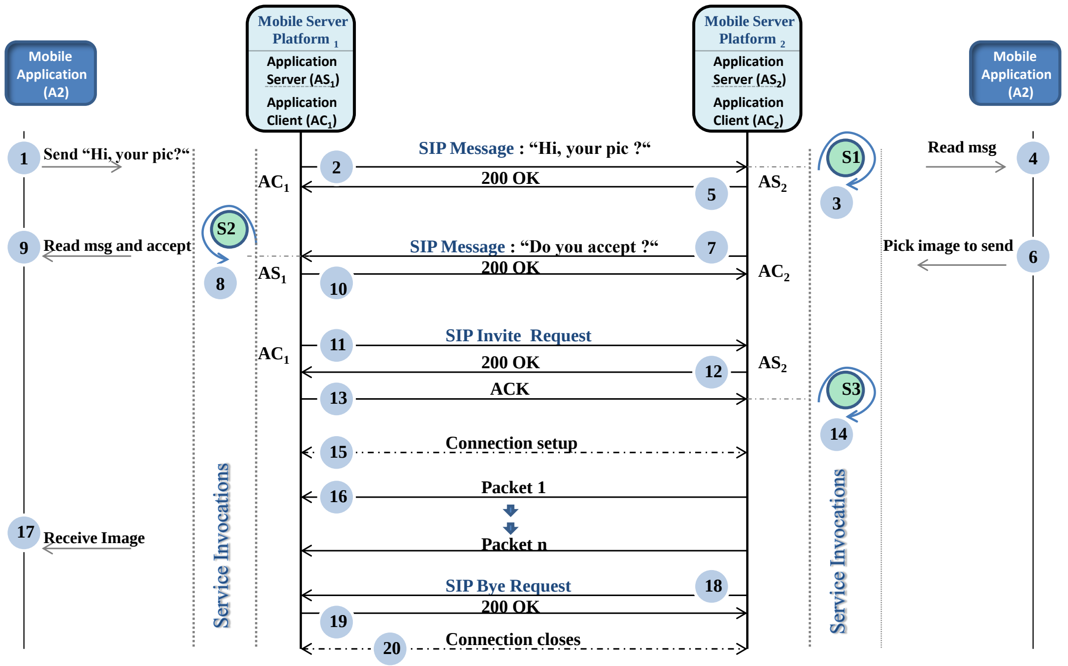



FIGURE 3.5: M2M Chat application scenario

Now, the question is where and how the IMS plays role in this scenario and helps the devices to keep direct contact over the operator network. This discussion could be divided into two main phases: Session establishment (SE) phase and media delivery (MD) phase.

**Session Establishment Phase** In the chat messaging example shown in Figure 3.5, the SE phase is comprised of steps 1-14. Here, simple message sharing is done in order to establish a media session without the actual media being transferred between two applications. These are the messages sent over the IMS network and go through the SIP proxies, the CSCFs. However, every single terminal is required to upload its current location (network address) before a session startup by registering to the IMS core network. Therefore, the communication between two mobile devices is provisioned via SIP proxies in the SE phase requiring no direct connection among them.

**Media Delivery Phase** The steps 14-17 denote the MD phase in Figure 3.5 which is solely responsible for actual media transfer among the communicating parties. The media is transferred through the operator's network gateway which involves no IMS SIP proxy in between sending and receiving devices. Therefore, a special mechanism is required here to make the devices directly contact over the operator's IP network. There are two possibilities to solve this problem. First, like every application server in the IMS network the operator should provide a direct access to the user devices acting as MobAS either by introducing special policies or by assigning public IPs to them. Second, operators should utilize full strength of IMS by shifting their system to IP V6 as described in [15] .

### 3.2.3 Quality of Service Control in the LTE - Evolved Packet Core

Multimedia services due to their delay-sensitive nature require special treatment over bandwidth-constrained cellular networks as compared to rich-bandwidth fixed IP networks. Therefore, in order to address the QoS issue indicated in section 3.1, the MSP utilizes the full strength of the 3GPP Policy and Charging Control (PCC) architecture as depicted in figure 3.3. Along with the IMS ecosystem, the PCC architecture provides an efficient mechanism to differentiate among different types of subscribers profiles and services data. Moreover, it facilitates the network operators to control the ongoing traffic in the network and perform appropriate actions to attain desired performance.

The PCC architecture is explained in 3GPP Release 8 for policy and charging control in the LTE-EPC[38], as shown in Figure 3.6. It is important to note that the IMS is an essential part of this architecture which is playing a critical role in the session establishment phase. Here, the IMS Core has the key responsibility to identify the session type, prepare QoS treatment and signal this information to the network gateway.

Figure 3.6 can be easily understood with a use case where a subscriber sets up an IMS voice call. The subscriber's SIP request comes to the CSCF in order to establish the session. The CSCF first checks whether the subscriber is authorized to this call and then transfer the request to the callee. Upon receiving an OK message by callee's CSCF, the CSCF sends the session information to the policy controller (PCRF) using an AAR (AA-Request) message. The PCRF in turn selects PCC rules from the policy database and prepares QoS information to be applied to this particular session. This information is then transfered to the network gateway in an RAR (Re-Auth-Request) message. Finally, the gateway establishes the network bearer according to the information received from PCRF.

For more details on how QoS guarantees can be achieved in the M2M service networks, please refer to chapter 6.

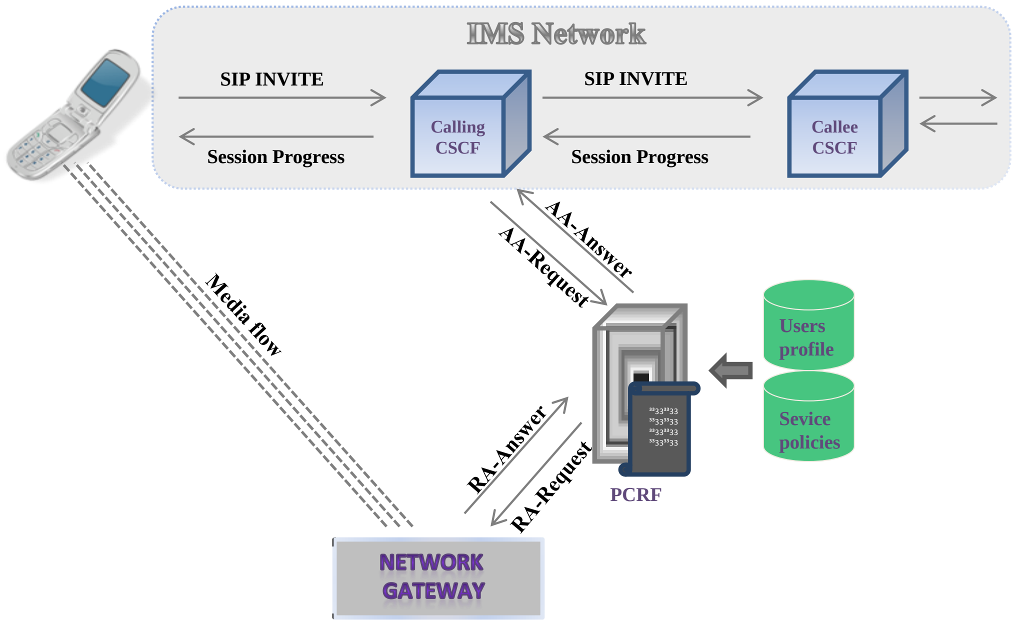



FIGURE 3.6: QoS Control in the LTE - Evolved Packet Core

## 3.3 Conclusion

The foundation of M2M service concept over the mobile cellular network is not straight forward: the proposal has to integrate a number of communication standards and protocols, address organizational interests and benefits, and to assure the QoS requirements for end users' satisfaction at the same time. It is worthy to mention that the former research of MSP [9] has made significant contribution in the space of M2M service provisioning. However, there is a lack of support for cellular network interfaces, protocols and policy and charging mechanisms required to realize M2M concept for telecom networks. Therefore, this chapter is dedicated to propose the required modifications and extensions in the existing MSP for standardized M2M service provisioning over the cellular operator network. The proposed solution is as follows:

- A novel concept of Mobile Application Server (MobAS) has been introduced in the operator's LTE / IMS network for hosting M2M multimedia services from the user handheld devices. Same as SIP ASs in the IMS network, the proposed MobAS has the ability to register itself in the IMS network and invoke multimedia sessions through standardized interfaces of the operator network, if the operator permits to do so.

- From accessibility point of view, every MobAS will be assigned a static IP address and port number in the DNS system of the operator network. However, in the current IMS testbed the assignment of IP addresses is a manual process, which could a be tiresome process for bigger networks. Therefore, it is recommended that IPs are assigned to the user profiles instead of individual devices.

- The SIP protocol mappings are created with the REST messaging framework of the MSP proposed in [9]. Some standardized M2M services such as text messaging, file sharing and multimedia audio/video streaming are developed as a part of extended framework. The architectural details of the proposed messaging and streaming framework will be covered in chapter 4 and chapter 5.

- In order to ensure QoS provisioning of M2M multimedia services, an M2M QoS framework (detailed architecture in chapter 6) is implemented following the 3GPP Policy and Charging Control (PCC) standard.

# Chapter 4

# Mobile-to-Mobile Multimedia Messaging Framework

This chapter enfolds the design and implementation of M2M messaging framework of the MSP. The proposed instant messaging services such as text messaging and file sharing support the standardized protocols and interfaces of the IMS network. The proposed architecture is aimed to conform to the functional and non-functional requirements of the Instant Messaging (IM) service addressed by the 3GPP telecom association. These requirements are summarized in section 4.1.1. Furthermore, the chapter provides the performance evaluation of the newly implemented services. The work presented in this chapter is mainly based on the author contribution in [21, 20]. Moreover, a master thesis guided by the author [57] has elaborated implementation details and conducted some additional measurements.

## 4.1 Instant Messaging in IMS

Instant Messaging (IM) is known as one of the most popular services among users to share content in near-real time. The content in an IM could be a text, an image, an audio or a video clip. 3GPP specifies the requirements for ASs and the IMS terminals to enable support for IM for information exchange. The IM service combines perfectly with the presence service as users know when the friends are online and can start messaging conversation [28, 3].

There are two modes of IM in the IMS: pager mode and the session-based mode. Pager mode came with the introduction of IMS in 3GPP release 5. Pager mode instant message is short message with limited data capacity and has no connection with any other messages. The pager mode contains the basic requirement for the application servers to be able to send short IM to the IMS terminal. Pager mode IM is similar to Short Message Service (SMS) in the cellular network. It uses the IETF extension which consists of a SIP method called MESSAGE. This MESSAGE method can carry any type of payload in the body of the message, if formatted with an appropriate MIME type [5].

Release 6 of the 3GPP specifications for the IMS introduced session based IM in the IMS network. SIP INVITE method is used to establish the session. The Message Session Relay Protocol (MSRP) is used to transport the messages in the IMS. The MSRP is implemented in the IMS terminals. MSRP runs over the media plane, which comprises of the operator's network gateway and involves no IMS SIP proxy in between sending and receiving devices. In other words, the MSRP messages need not traverse SIP proxies. This is a benefit since SIP proxies are not bothered with proxying large instant messages.

### 4.1.1 Instant Messaging Requirements in IMS/LTE Network

The 3GPP specification for IM defines some implementation requirements for the pager-mode and session based IM. In the scope of the dissertation, some key requirements are identified and discussed below. However, the complete set of 3GPP requirements can be found in 3GPP TS 22.340 [5].

***Interoperability*** The network operators have different network configurations and commercial requirements, hence, IMS messaging shall be supported in a manner to meet the IMS requirements of the operator and thus the standardized set of functionalities must ensure interoperability across different networks and terminals supporting IMS messaging.

***Differentiation*** There shall be a mechanism to differentiate session based messaging from other messaging types. The functionalities included in session based messaging must ensure interoperability between different terminals and networks.

***Terminal capabilities*** When including the participants in the messaging session the terminal capabilities must be taken into account like the display capabilities, media content types and media content formats supported by the device, media storage capacity and the encryption and security mechanisms supported by the terminal.

***Session properties*** The subscriber should be able to request the message session properties.

***QoS control*** The network operator providing the IMS messaging service shall be given the possibility to choose, wherever possible, the QoS parameters and also be able to enforce the preferred transport mechanism and parameters both for the UE originated and UE terminated messages.

***Messaging content*** Regarding the message content, 3GPP specifies that the content size shall not be limited and it should be possible to carry different media including text, audio, video and images and the media types shall be MIME encoded. Session based messaging shall use a minimum set of supported formats to ensure interoperability between different networks and terminals and these formats shall be common to all IMS messaging types. The content formats shall be aligned to enable interworking with the 3GPP and Internet messaging solutions. It shall be possible to compose a message of multi-media (e.g. voice and video).

***Enable and disable messaging*** IMS messaging shall be able to support a request from the service provider to enable or disable message delivery and submission and similarly, it shall also support a request from the user to enable or disable message delivery and submission.

***Customization*** The user shall be able to manage his service profile related to the IMS messaging that means the user can customize his messaging environment within the capabilities of the messaging application, terminal and the network.

***Acknowledgments*** The messages are transported by the IMS network to the recipient's terminal and the sender can receive the service delivery acknowledgments for the messages sent by him.

## 4.2 Realization of M2M Instant Messaging

This section explains how the Mobile Server Platform (MSP) is utilized to realize an M2M instant messaging over the operator IMS network. Imagine the chat messaging session between two mobile users, Bob (A1) and Alice (A2), as discussed in section 3.2.2.3. As Figure 3.5 depicts that both users exchange a couple of short SIP messages until Bob wants Alice to send him a picture. Alice picks a picture from the picture gallery and presses Ok to send. In response to OK, the Alice's MSP sends a short SIP message to Bob's MSP in order to have its permission to receive an image file. After receiving a 200 OK message from Bob's MSP, Alice MSP creates a SIP INVITE session with Bob's MSP and the actual image transfer starts.

In the above scenario, the communication between two end points can be categorized in two different forms: short term messages (like pager mode IM) such as text messaging in the start of the conversation and long term messages (like session based IM) such as multimedia file transfer. Hence, the following sections enfold the implementation of these two messaging types.

### 4.2.1 Text Messaging

As mentioned earlier that MSP supports two different types of service invocation mechanisms: synchronous and asynchronous. The text messaging service is implemented by synchronous invocation mechanism of MSP. The service allows an IMS user to send simple chat messages to other registered users in the network. It only requires the message content and the destination address in order to send a message to the destination user. The user is notified after the message is successfully sent.

Hence, in case of any SIP MESSAGE received by the MSP, a synchronous service is invoked which reads the message and sends a 200 OK message in response. In general, any kind of SIP request that follows a request-response approach can be implemented by the synchronous mechanism of MSP.

Figure 4.1 depicts the signaling flow of SIP MESSAGE request from one MSP to another. Whenever a SIP Message reaches an IMS network, two major operations are performed by the network entities. First, a Diameter request-response operation is performed by I-CSCF in order to verify the subscription of the mobile user. The verification is done by HSS database. The second major operation that is performed by S-CSCF is called Initial Filter Criteria (IFC) evaluation. The IFC evaluation is done in order to locate the respective application server responsible for handling the coming service request. The IFC is the grouping between a Trigger Point (the logical expression matching a message) and an Application Server. The absence of a Trigger Point in an Initial Filter Criteria indicates that the message should always be forwarded to the respective Application Server.

Listing 4.1 shows a SIP MESSAGE method which is sent by Alice to Bob. The text message is included in the content field of the SIP message. The public ID of the destination user (Bob's MobAS) is added in the "To"header field. The IMS proxy server translates the IP address of the destination user against its public ID and forwards it to particular MobAS. The REST URL is also included as SIP header, which points to the destination service in the MobAS.

After the SIP MESSAGE is received by Bob's MobAS, the message is parsed and the REST URL is extracted. The message is then forwarded to the text messaging service. The service reads the content of the message and informs the Bob's MobAC. The service then contacts the Response Handler (see figure 3.3). The Response Handler creates a 200 OK message and sends back to the Alice's MobAC. The SIP 200 OK message is shown here in listing 4.2.

```
MESSAGE sip:greetings@bob-server.com SIP/2.0
Max-Forwards: 69
CSeq: 1 MESSAGE
resourcemethod: GET
P-Called-Party-ID: <sip:greetings@bob-server.com>
Content-Length: 84
Call-Info: <http://www.antd.nist.gov>
rest-service-uri: POST/IMsync
Contact: "alice" <sip:alice@10.0.2.15:5070>
Record-Route: <sip:127.0.0.1:5082;from-tag=12345;lr>
P-Asserted-Identity: <sip:alice@ericsson.com>
To: "Bob" <sip:bob@ericsson.com>
From: "Alice" <sip:alice@ericsson.com>;tag=12345
Call-ID: 79a6e89a0f8c55849222730b2a90769f@10.0.2.15
Via: SIP/2.0/udp 127.0.0.1:5082;branch=
    z9hG4bKaba7c1f00b8133b045de689d1ca1b46e,
SIP/2.0/UDP 10.0.2.15:5070;branch=
    z9hG4bK29cb37aa1e68dd24e29b2ca0762fae9a383639;
```

```
received_port_ext=5081;received=192.168.1.4
Content-Type: text/xml
<?xml version='1.0' ?><RESTRq><Rq><demo>Hi Bob, Alice here!</demo></Rq
    >
</RESTRq>
```

Listing 4.1: SIP MESSAGE from Alice to Bob

```
SIP/2.0 200 OK
Contact: "Alice" <sip:alice@192.168.1.4:5061>
Content-Type: text/xml
CSeq: 1 MESSAGE
Via: SIP/2.0/UDP 127.0.0.1:5082;branch=
    z9hG4bKaba7c1f00b8133b045de689d1ca1b46e;
received=10.0.2.2,SIP/2.0/UDP 10.0.2.15:5070;branch=
    z9hG4bK29cb37aa1e68dd24e29b2
ca0762fae9a383639;received_port_ext=5081;received=192.168.1.4
Content-Length: 79
Call-ID: 79a6e89a0f8c55849222730b2a90769f@10.0.2.15
From: "Bob" <sip:bob@ericsson.com>;tag=12345
To: "Alice" <sip:alice@ericsson.com>;tag=888
```

Listing 4.2: SIP 200 OK

### 4.2.2 Multimedia File Sharing

The M2M file sharing service is implemented by the asynchronous invocation mechanism of the MSP. In contrast to a SIP MESSAGE request, SIP INVITE request is used to create a particular session for a multimedia data transfer. Figure 4.2 depicts how a SIP INVITE request goes from one MSP to another over the operator IMS network. A SIP INVITE request reaches to MSP in a similar way as SIP MESSAGE request. However, after receiving an INVITE request, the MSP completes a three way handshaking procedure. And, the actual multimedia transfer is done via MSRP protocol, which is a standard for session based Instant Messaging introduced by 3GPP.

In case of receiving a SIP INVITE request the MSP sends a 200 OK response to sender and waits for an ACK before it invokes an asynchronous service. This is exactly the point for which the initial asynchronous service design depicted in 3.2 is

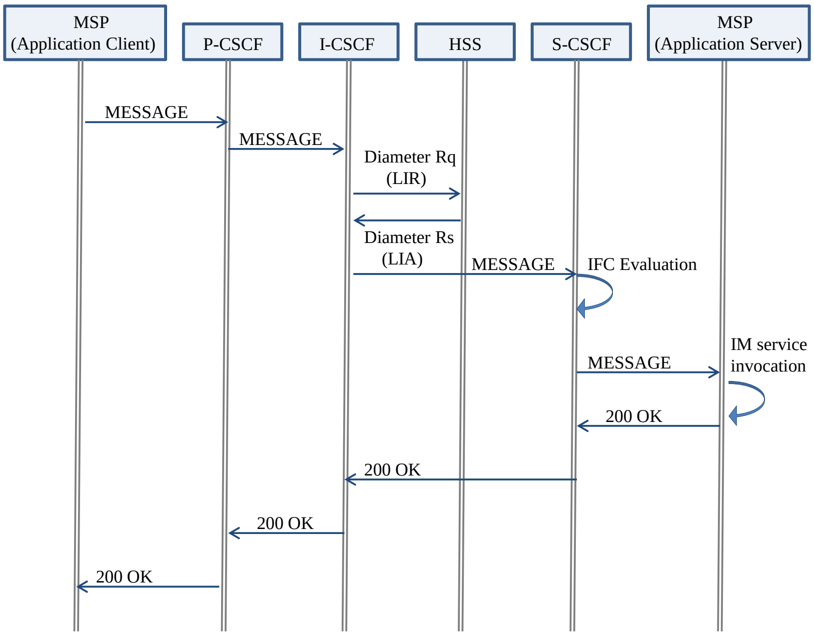



FIGURE 4.1: SIP MESSAGE Flow

extended in a way to support an additional notification message (ACK) from client to server after an INVITE request is sent and response (200 OK) is received.

The discussion of the asynchronous file sharing service is further divided into two major phases: session establishment phase and media delivery phase.

#### 4.2.2.1 Session Establishment Phase

The SIP INVITE request is sent by Alice with the public SIP address of the destination MobAS. In this case it would be bob-server@ericsson.com. This SIP INVITE request message reaches the IMS proxy server CSCF. The CSCF checks the DNS and finds the IP against the requested server "bob-server.ericsson.com". The SIP INVITE request is then forwarded to that address. The SIP INVITE method also contains the contact header in which the IP and port is specified where 'Alice' wants to receive the response. A sample SIP INVITE is shown in the listing 4.3.

```
INVITE sip:greetings@bob-server.com SIP/2.0
Max-Forwards: 69
```

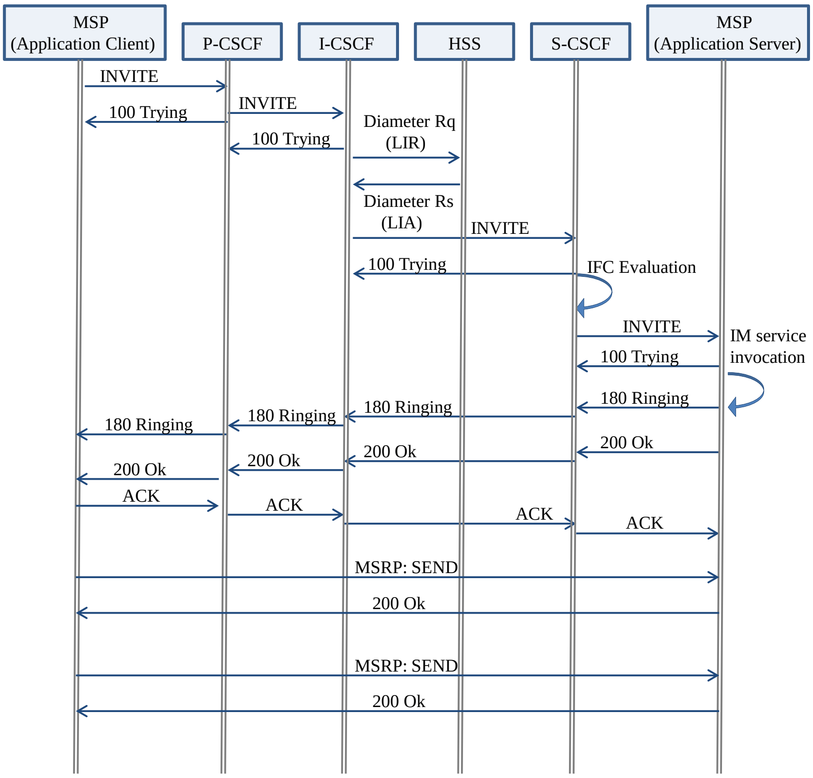



Figure 4.2: SIP INVITE Flow

```
CSeq: 1 INVITE
resourcemethod: GET
P-Called-Party-ID: <sip:greetings@bob-server.com>
Content-Length: 751
Call-Info: <http://www.antd.nist.gov>
rest-service-uri: POST/Request-Response/Factory/createInstanceRq/
    IMasync/add
Contact: "Alice" <sip:alice@10.0.2.15:5071>
Record-Route: <sip:127.0.0.1:5082;from-tag=12345;lr>
P-Asserted-Identity: <sip:alice@ericsson.com>
To: "Bob" <sip:bob@ericsson.com>
From: "Alice" <sip:alice@ericsson.com>;tag=12345
my-other-header: my new header value
Call-ID: 37c08f9666a687a3027ac22b98483158@10.0.2.15
Content-Type: application/sdp
Via: SIP/2.0/udp 127.0.0.1:5082;branch=
    z9hG4bK4a2b443a5792a53b7d13c83662dd87cf,
SIP/2.0/UDP 10.0.2.15:5071;branch=
    z9hG4bK3054372ee7534b8d107a936cd29931e1393030;
received_port_ext=5081;received=192.168.1.4
```

```
v=0
o=coco 1326930823911 1326930823911 IN IP4 alice@ericsson.com
s=sdp offer (recvonly)
c=IN IP4 alice@ericsson.com
t=0 0
m=message 9097 TCP/MSRP
a=recvonly:recvonly
a=accept-types:message/cpim
a=accept-wrapped-types:*
a=path:msrp://alice@ericsson.com:9097/jshA7we;tcp
a=file-name:sample.jpg
a=file-transfer-id:aCQYuBRVoUPGVsFZkCK98vzcX2FXDIk2
a=file-size: 38503
```

Listing 4.3: SIP INVITE from Alice to Bob

***SDP Offer-Answer Model :*** The two parties sharing multimedia file need to agree on a set of parameters before the actual file transfer. This is done using the SDP offer-answer model. In actual sessions, any media type can be transferred but it needs to be specified in advance with the help of the SDP message. The SDP exchange requires the existence of SIP which is capable of carrying the SDP messages between two devices for the purpose of session establishment between them. The attributes like file name, file size, file type are included in the SDP. Protocol operation starts when a user sends SDP offer to another user. The user receiving the offer may generate an answer or simply reject the offer. Hence, if the requested file is available in the database or sdcard of the user device, the answer is generated. In case the requested file is not available, then the request is rejected.

Listing 4.4 shows a sample SDP offer message. The 'm' field describes the media stream type and the 'a' fields following the 'm' line further describes the attributes related to that media stream. In the given example, 'm' field indicates that the data type is MSRP message and will be transferred using a TCP connection. Every file has a file transfer id which is included with all the chunks of the same file, hence, file transfer id is also included. For MSRP communication, the IP and the port number of the destination is required. This information is described in a MSRP URL. This MSRP URL is also included in the SDP message. The file name and the file size attributes further describe the file to be transferred. The support for SDP is provided in the framework using the existing standard JAINSIP library for SDP "javax.sdp". Listing 4.5 shows a code snippet to generate a simple SDP message.

```
v=0
o=coco 1326930823911 1326930823911 IN IP4 alice@ericsson.com
s=sdp offer (recvonly)
c=IN IP4 alice@ericsson.com
t=0 0
m=message 9097 TCP/MSRP
a=recvonly:recvonly
a=accept-types:message/cpim
a=accept-wrapped-types:*
a=path:msrp://alice@ericsson.com:9097/jshA7we;tcp
a=file-name:sample.jpg
a=file-transfer-id:aCQYuBRVoUPGVsFZkCK98vzcX2FXDIk2
a=file-size: 38503
```

LISTING 4.4: An SDP Offer Message

```
sessionDescription = sdpFactory.createSessionDescription();
Version v = sdpFactory.createVersion(0);
SessionName sn = sdpFactory
.createSessionName("sdp offer (recvonly)");
Origin o = sdpFactory.createOrigin("coco",
System.currentTimeMillis(), System.currentTimeMillis(),
"IN", "IP4", host_ip);
sessionDescription.setVersion(v);
sessionDescription.setOrigin(o);
// sessionDescription.setInfo(i);
sessionDescription.setSessionName(sn);
sessionDescription.setConnection(sdpFactory
.createConnection(host_ip));
Vector<MediaDescription> mediaList = new Vector<MediaDescription>();
MediaDescription md = sdpFactory.createMediaDescription("message",
port, 1, "TCP/MSRP", new int[0]);
md.getMedia();
md.setAttribute("recvonly", "recvonly");
md.setAttribute("accept-types", "message/cpim");
md.setAttribute("accept-wrapped-types", "*");
md.setAttribute("path", msrp_uri);
md.setAttribute("file-name", "proposal.pdf");
md.setAttribute("file-transfer-id", ftID);
mediaList.add(md);
```

```
sessionDescription.setMediaDescriptions(mediaList);
```

Listing 4.5: Code snippet to generate a simple SDP message

Bob's MobAS receives the SIP INVITE message. The SIP headers are then parsed. The REST URL is parsed to extract the target service. A thread of the file sharing service is created but not started. If the service exists and the parameters mentioned in the SDP header fields are agreed upon, then a SIP 200 OK response is sent back. The Restful Response Handler is used to create a SIP 200 OK response. The SIP 200 OK is then sent to the Alice via the proxy servers CSCF. Listing 4.6 shows a sample SIP 200 OK response message.

```
SIP/2.0 200 OK
CSeq: 1 INVITE
Content-Length: 468
Record-Route: <sip:127.0.0.1:5082;from-tag=12345;lr>
Contact: "Bob" <sip:bob@192.168.1.4:5060>
To: "Bob" <sip:bob@ericsson.com>;tag=888
From: "Alice" <sip:alice@ericsson.com>;tag=12345
Call-ID: 37c08f9666a687a3027ac22b98483158@10.0.2.15
Via: SIP/2.0/UDP 10.0.2.15:5071;branch=z9hG4bK3054372ee
7534b8d107a936cd29931e1393030;
received_port_ext=5081;received=192.168.1.4
Content-Type: text/xml
v=0
o=bob 1326930826199 1326930826199 IN IP4 bob@ericsson.com
s=sdp answer (sendonly)
c=IN IP4 bob@ericsson.com
t=0 0
m=message 59418 TCP/MSRP
a=sendonly:sendonly
a=accept-types:message/cpim
a=accept-wrapped-types:*
a=path:msrp://alice@ericsson.com:59418/9di4ea;tcp
a=file-name:sample.jpg
a=file-transfer-id:aCQYuBRVoUPGVsFZkCK98vzcX2FXDIk2
```

Listing 4.6: SIP 200 OK message

Listing 5.5 shows a SIP ACK message. Upon receiving the SIP ACK message the SIP listener forwards the request to the Restful Handler. The SIP service thread is then started and the file transfer begins.

```
ACK sip:bob@192.168.1.4:5060 SIP/2.0
Contact: "Alice" <sip:alice@10.0.2.15:5071>
CSeq: 1 ACK
Via: SIP/2.0/udp 127.0.0.1:5082;branch=z9hG4bKf391367a
d35fe5f9f2a0eedbc6509d4e,SIP/2.0/UDP 10.0.2.15:5071;
branch=z9hG4bK153ff16e9db7f0dab5d749bf709250cb393030;
received_port_ext=5081;received=192.168.1.4
Max-Forwards: 69
Content-Length: 0
Call-ID: 37c08f9666a687a3027ac22b98483158@10.0.2.15
From: "Alice" <sip:alice@ericsson.com>;tag=12345
To: "Bob" <sip:bob@ericsson.com>;tag=888
```

LISTING 4.7: SIP ACK message

#### 4.2.2.2 Media Delivery Phase

The service starts only after the ACK is received. For actual media transfer, a TCP connection is setup between the two devices. The service fetches the file, decomposes the file into small chunks. These chunks are then sent as MSRP messages over the TCP connection. MSRP header has a field that shows if it is the last packet or not. Thus the Bob's MobAC knows when the last MSRP message is received. After successfully receiving all the data chunks, they are combined to get the file. The session is terminated after the successful transfer of the MSRP messages using the SIP BYE request.

### 4.2.3 MSRP Implementation

The MSRP is a protocol used in the IMS framework to transfer multimedia data in session based IM as defined in 3GPP standards. MSRP defines two types of request or method: 1) SEND Request 2) REPORT Request

***SEND Request:*** SEND requests are used to deliver a message to the other party. A message could be a complete message or a part (chunk) of a complete message.

***REPORT Request:*** A REPORT request reports on the status of a previously sent message.

In contrast to the individual messages sent in the pager-mode IM, the session based IM contains a series of exchange of messages with a definite start and a definite end. Session based IM has quite many advantages over the pager-mode IM including tighter integration with different media-types, direct client to client communication, and brokered privacy and security.

Several open-source libraries for implementation of the MSRP are found. But no proper documentation or tutorial is available on the use of these MSRP libraries. Therefore, a customized functionality of the MSRP with respect to the file sharing scenario is implemented from scratch. The written MSRP class follows the IETF standard [29] in order to achieve the following functionalities:

- Converting the byte array of the given file data to small chunks of fixed size.
- Adding MSRP headers to the individual chunks depending on the chunk number.
- Getting a received chunk information included if the chunk is the last one or not.
- An MSRP parser in order to read an MSRP message and extract the payload.

A typical MSRP request contains an MSRP URI, which is used to identify a session of instant messages at a particular MSRP device, as well as to identify an MSRP relay in general. An MSRP URI follows a scheme as define in Reference RFC 3986 [25], which holds the IP/ domain name and port, the session identifier, the transport method and additional optional parameters. Listing 4.8 shows a sample MSRP SEND request. Here msrp://alice@ericsson.com:9097asaO6sdkht;tcp is an example URI, where:

- msrp is the scheme, it can also be msrps
- alice@ericsson.com is the domain
- 9097 is the port where it want to get the requests on the IP of the above domain
- asaO6sdkhto is the session identifier
- tcp is the transport.

```
MSRP d93kswow SEND
To-Path:msrp://bob@ericsson.com:9097/9di4eae923wzd;tcp
From-Path:msrp://alice@ericsson.com:7654/iau39soe2843z;tcp
Message-ID:fk3tv7ip3tji
Byte-Range:1-750/169453
Content-Type: message/cpim
To:Bob <sip:bob@ericsson.com>
From:Alice <sip:alice@ericsson.com>
Content-Type: text/plain

iVBORw0KGgoAAAANSUhEUgAAABQAAAAiCAYAAABfqvm9AAAABGdBTUEAAK/I
NwWK6QAAAAlwSFlzAAAOxAAADsQBlSsOGwAABG9JREFUSEulVm1MW2UUfvrJ
AhQCpX5QaImwlW8NMJgQEgkJi4hsUVg0hkhM4Jf8mNkPjAlOwR8S+aH+MRij
yYz7AQEjwqIDZFFnS5nQQURGB0NYy0e7rV90QOF63tuBXGgpuNPe5Oa95zz3
Oec873mviCNDAJubm0N/fz9MJhOcTifvERkZiezsbJSWliI5OTlQGMAAd5vH
4+Gampo4lUrFRchkXP7TT3Kv6pK5KrpOxT/FRcplXGxsLNfY2MjRi/aGc6Ld
DGdnZ1FVVQXrrSm8c/JAuroKEAi9rPZ3ILV6ULHpBmfDI8iSqNFZ2cn
UlNTd9juAJrNZpSUClBdKsJ3Z09DyYA2fASyKUyNgctkcDjdeLPnKv5wejAw
MIDMzEjwf0er0oKirCEw9s+OHcGcilFOTbA7S3YlIJNqn81R29mJTIYTAY
EBUVBT6X1tZWLE/fwqWzLx4OjAXRCyUQ4eszp7G+aEFKCnBe
p8X5onzg4Vrg7gVbPRaGL4fH8N7IOM9SPDQ0BPuiFedSjwPrG4Iwi9uDi78N
I+Ory/zF7tmawCjmldQU+FxOvpaSpKSkix0NC6cygW2tgS+bfTmD343YmXV
y1/X5i
--------d93kswow+
```

Listing 4.8: Sample MSRP message chunk generated

#### 4.2.3.1 MSRP Chunking

The MSRP class gets the complete message. It creates the chunks itself. The MSRP message chunking is explained here by considering a small data of 148 bytes being divided into a chunk size of 137. The message is divided into two chunks. The first chunk is shown in listing 4.9. The string data "ABCD" is sent in the shown MSRP chunk. The second and the final chunk is then sent with String "1234567890" as shown in listing 4.10.

```
    MSRP d93kswow SEND
    To-Path: msrp://bob@ericsson.com:8888/9di4eae923wzd;tcp
    From-Path: msrp://alice@ericsson.com:7654/iau39soe2843z;tcp
    Message-ID: 12339sdqwer
```

```
Byte−Range: 1−137/148
Content−Type: message/cpim

To: Bob <sip:bob@example.com>
From: Alice <sip:alice@example.com>
DateTime: 2006−05−15T15:02:31−03:00
Content−Type: text/plain

ABCD
−−−−−−−d93kswow+
```

Listing 4.9: MSRP message first chunk

```
MSRP op2nc9a SEND
To−Path: msrp://bob@ericsson.com:8888/9di4eae923wzd;tcp
From−Path: msrp://alice@ericsson.com:7654/iau39soe2843z;tcp
Message−ID: 12339sdqwer
Byte−Range: 138−148/148
Content−Type: message/cpim
1234567890
−−−−−−−op2nc9a$
```

Listing 4.10: MSRP message chunk N

As the complete message is taken into account by the MSRP class, so the CPIM headers are sent only in the first MSRP message. The "Content-Type" and the "Byte-Range" headers are present in both chunks.

#### 4.2.3.2 Base64 Encoding

In order to send a file over the network, the file is converted to a byte array for processing. The byte array is divided into chunks and Base64 encoding is applied on each chunk. The Base64 is an encoding scheme that represents the binary data in an ASCII string format. It is done by translation into radix-64 representation. Base64 scheme is more popular in cases when the binary data needs be stored and transferred over the media designed to deal with textual data [58]. By fixing the chunk size the complete byte array is divided into n chunks.

n = filesize/chunksize + 1

Keeping the chunk size equal to 720 bytes, the byte array divided into 'n' number of smaller byte arrays. Then MSRP headers are added to each chunk. The MSRP header contains the information like message ID, session ID, content-type, byte range out of the total bytes to be sent and the information whether this is the last chunk or more chunks are to follow. After the successful chunking of the data into n number of MSRP chunks, the service is informed that the chunks are ready to be transferred.

#### 4.2.3.3 MSRP Parser

The MSRP parser gets the MSRP message. The MSRP headers are parsed here and the content is processed accordingly. The headers are removed and the media content is extracted from the message. The MSRP parser distinguishes a MSRP message chunk of one file from chunks of other transactions by the message identifier. It knows from the headers when it receives the last chunk. The last byte of the MSRP message has a value '+' or '$'. The '+' indicates, more MSRP chunks to follow and '$' indicates that this is the last MSRP chunk.

#### 4.2.3.4 MSRP Response

Whenever a MSRP message is successfully received, the parser informs the Response handler. Hence, the Response handler creates a 200 OK message and sends it back to the sender. The sender waits for the 200 OK messages and sends the next MSRP message only after it has received the MSRP 200 message. In case of any fault occurred in the transfer, the sender gets a Transaction Timeout Exception, so the previous MSRP chunk is sent again.

#### 4.2.3.5 MSRP Ordering

A method is also written in the MSRP class to combine all the chunks and return the complete message to the server.

## 4.3 Performance Evaluation

The performance evaluation of the proposed framework has been extensively carried out in order to study the processing latencies during the session establishment and multimedia transfer phase. The testbed environment consists of two user terminals and one server machine. The configuration of the testbed is as follows:

**User Terminal -1 with MSP**
Model: HTC Legend
Platform: Android 2.2
Camera: 5-megapixel color camera with auto focus and flash
Memory: ROM: 512MB RAM: 384MB
CPU Speed: 600MHz

**User Terminal -2 with MSP**
Model: Samsung Galaxy S3 (i900)
Platform: Android 4.1.2 (Jelly Bean)
Camera: 6-megapixel color camera with auto focus and flash
Memory: ROM: 16 GB RAM: 1 GB
CPU Speed: 1.4 GHz

**Server Machine with IMS Testbed**
Ericsson Service Development Studio is used as IMS testbed. It comes as an Eclipse plug-in. It is running on the machine with the following specifications.
System Manufacturer: Acer
Model: Aspire 5551G
Processor: AMD Phenom II, X3 Mobile Processor
Speed: 2.1 GHz
Memory: RAM: 4GB
Operating System: Genuine Windows 7 Home Premium 64-bit.

**Network Configurations**
ADSL: G.992.5 (ADSL2+)
Download: 17393 Kbps

Upload: 1149 Kbps
No of clients: 7

### 4.3.1 Session Establishment Latencies

Figure 4.3 shows a three way hand-shaking process to complete a session establishment phase of an asynchronous service. The processing latencies in this phase are computed at four different stages:

1. Processing latencies at Application Client (Tc)
2. Processing latencies at Application Server (Ts)
3. Processing latencies at IMS Proxies (Tp)
4. Network latencies Tn

The average time computed for session negotiations is about 1.8 seconds. So, the actual media transfer start after 1.8 seconds of the initial service request. Figure 4.4 depicts the individual times computed at each stage. It is noteworthy that the processing time at IMS proxy is very less, whereas most of the time is consumed by the application server in order to perform service invocation and response generation.

### 4.3.2 File Transfer Latencies

The different multimedia types such as text, image and audio files of different sizes were tested to evaluate multimedia transfer times. All the files types except the text files were encoded using Base 64 encoding scheme. Figure 4.5 shows image files of different sizes, before and after encoding. The blue bars indicate the number of bytes before encoding and the red bard indicate the number of bytes after encoding. After encoding the byte size increases by about 36 %.

The transfer rates for different image files are calculated and plotted in figure 4.6. It is observed that the file transfer rate starts with a reasonable value of around 26 kB/s for smaller files but keeps decreasing gradually with increase of file size. This

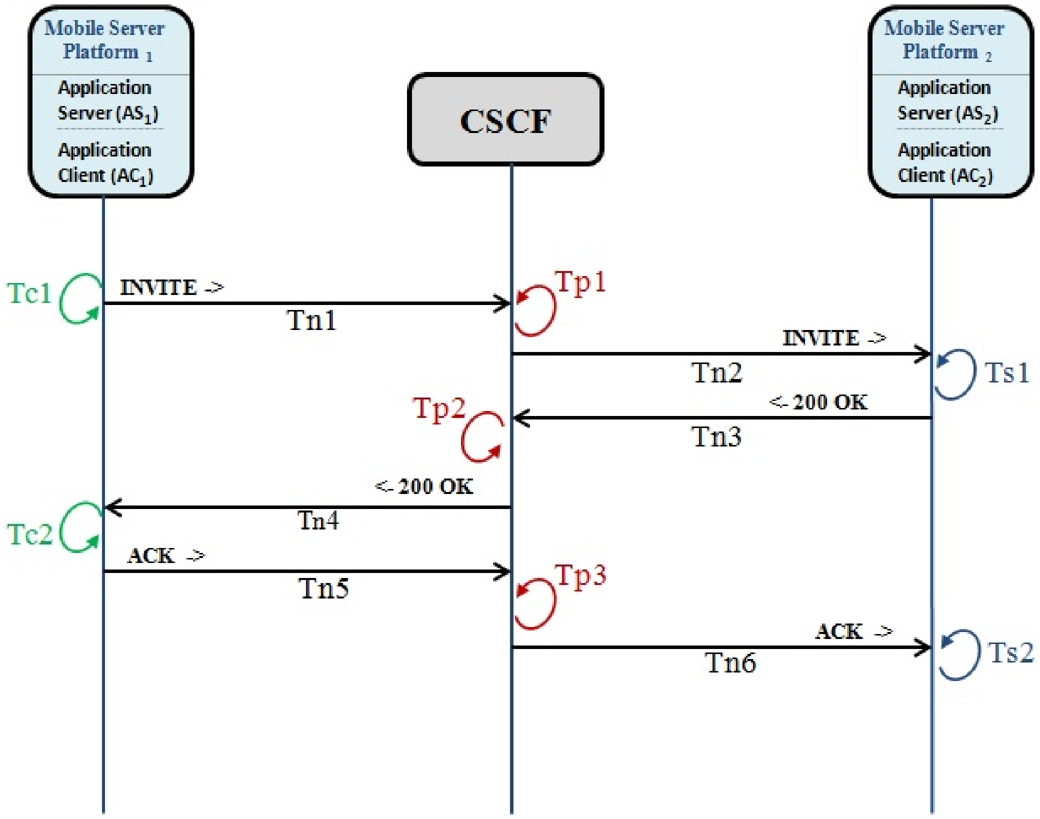



FIGURE 4.3: Classification of processing latencies in session establishment phase

is because of the fact that slow processing of the chunks at the android device slows down the file transfer speed.

Similarly, the file transfer times were also calculated for audio and text files of variable sizes as shown in figures 4.7 and 4.8. It is observed that the transfer rate increases rapidly, if the chunks received at the client side are not processed and stored.

The variable processing times increase with the increase of the file size. The media file is broken down into MSRP chunks of fixed sizes and transferred over the TCP link. All the received TCP segments are stored and added in the previously received segments. In case of bigger files, the received string becomes quite large. The memory heap size allocated to the android application is exceeded. The additional memory is allocated to the application on runtime and this produces delays in the file transfer. A new data chunk is not processed unless the previous chunk is stored successfully. In this case, the bottleneck is the processing speed and memory size

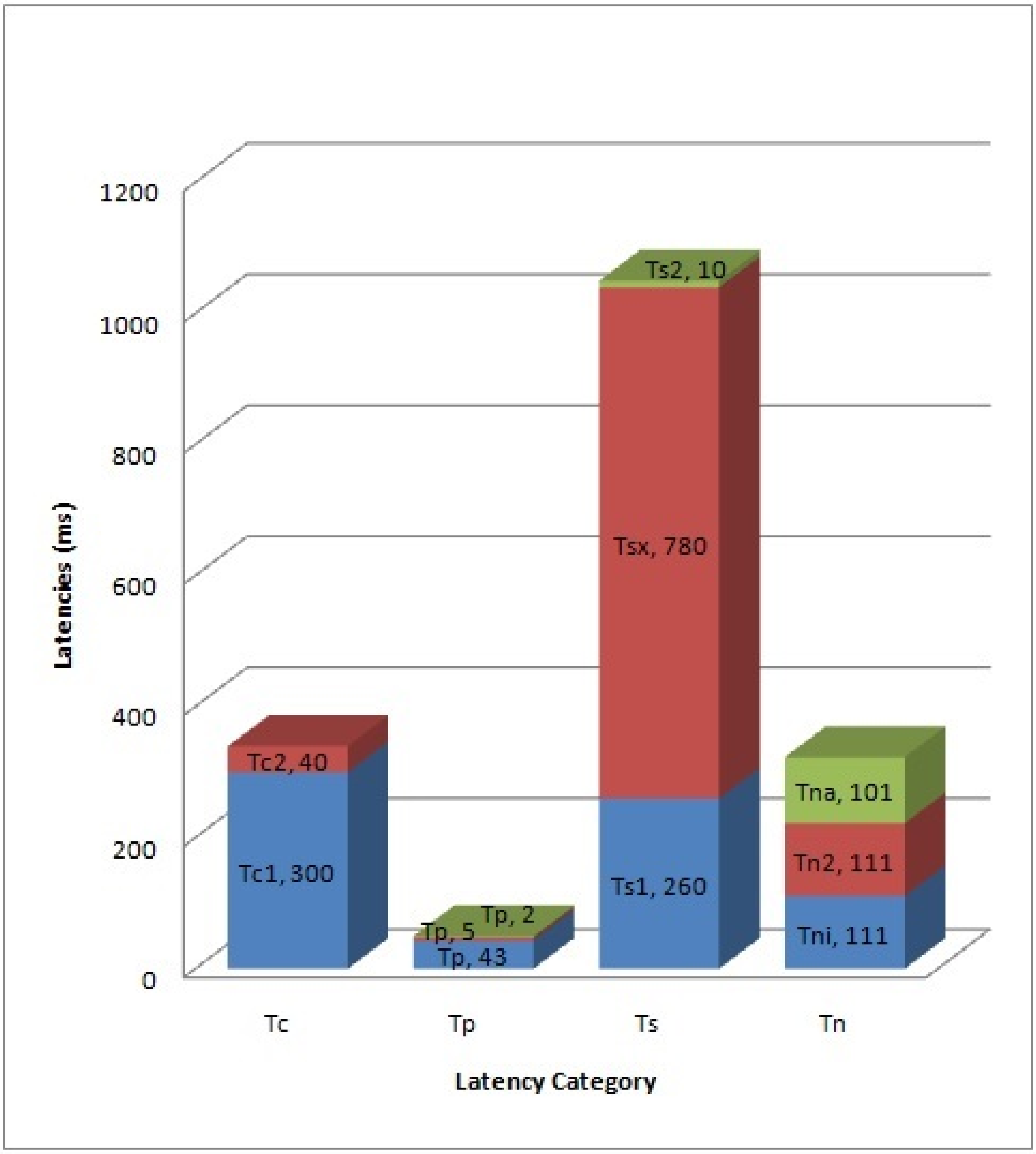



FIGURE 4.4: Session establishment latencies

and not the network speed as compared to the high end servers with high processing powers and huge memory sizes.

## 4.4 Discussion

Based on the calculations for session establishment latencies and multimedia content sharing times for session-based IM, some interesting results are discussed here. The session establishment latencies were independent of the file size and dependent on

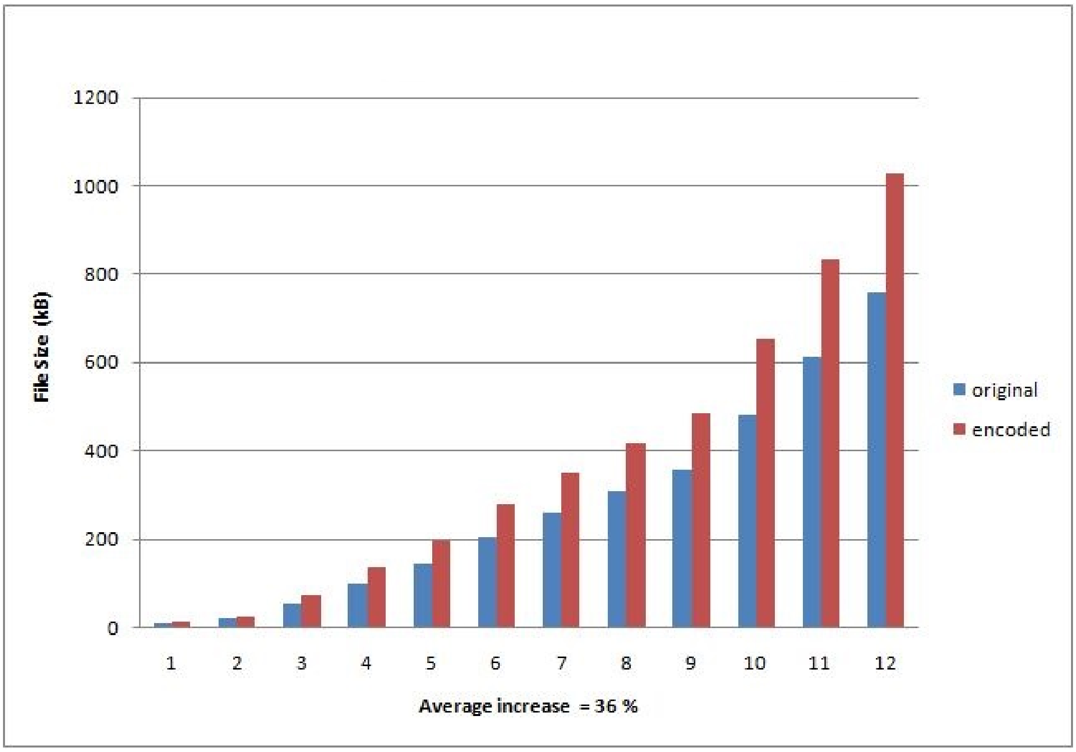



FIGURE 4.5: File size after Base64 encoding

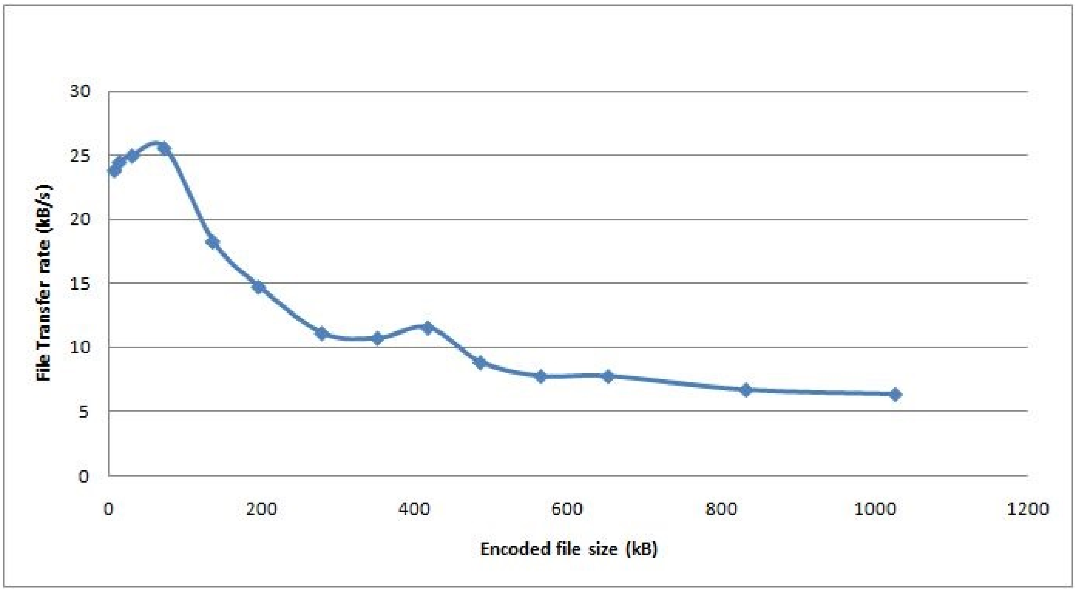



FIGURE 4.6: Transfer rate of image files of different sizes

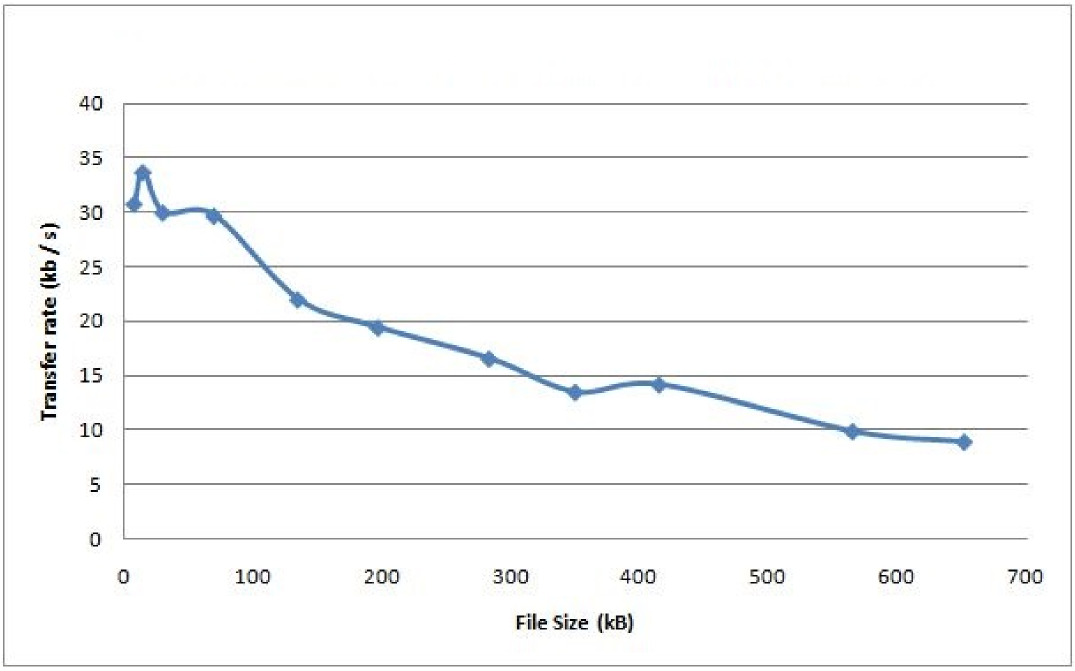



Figure 4.7: Transfer rate of audio files of different sizes

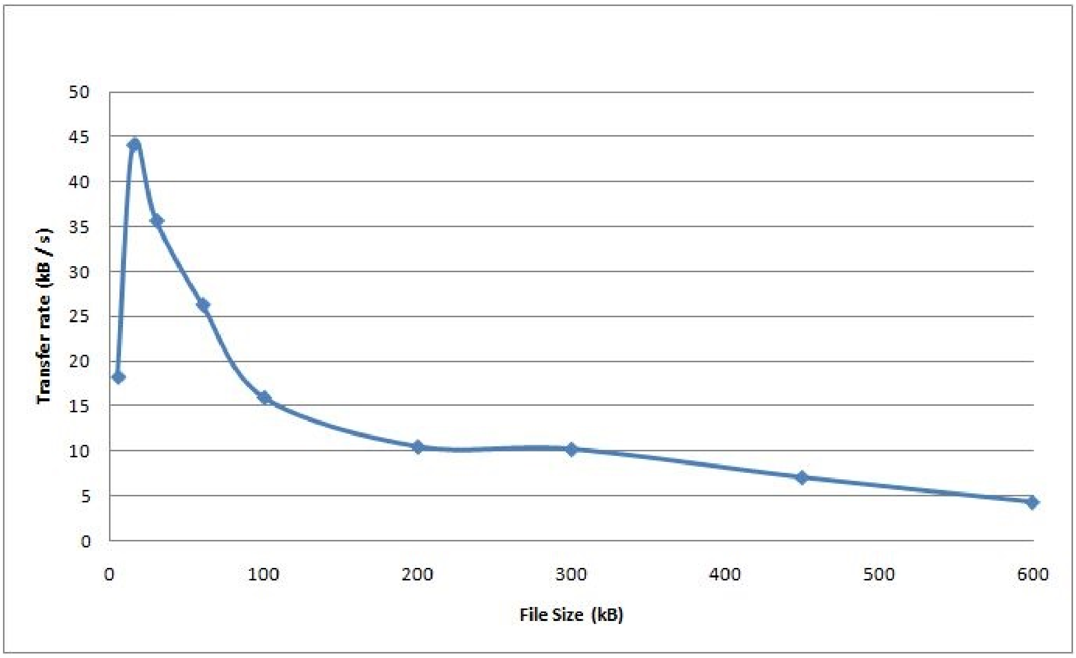



Figure 4.8: Transfer rate of text files of different sizes

the IMS network, MAS and IMS proxy server response times. It took fixed time around 1 - 1.5 second to establish a TCP session for multimedia sharing. After the session is established, the actual file sharing starts. The average data rates obtained for the initial chunks are around 30 kbps. It has been observed that with the increase of file size, the response becomes more unstable. The initial chunks are delivered at a rate faster than the later part of the multimedia file. This is due to the processing capabilities and the memory of the android device where the android application runs out of pre-allocated memory and tries to free more memory on runtime. Additionally, the time to save every incoming received chunk and appending it to the already received chunks increase as more number of chunks are received. In contrast, it was seen in servers running on high end machines that the file transfer rate remains relatively constant. This is due to the fact that the high end machines have the high processing power and sufficient memory.

In conclusion, it is deduced that the transfer speeds are affected by the limitations of the processing speeds and the memory size at the android server terminal. The processing of bigger files and storing them in the memory requires the resources that are limited in mobile server when compared to high end server machines. The file transfer speeds obtained vary between 25 Kbps to 7 kbps depending on the file size.

## 4.5 Conclusion

This chapter outlines the design of M2M multimedia messaging framework for the IMS-Mobile Server Platform proposed in chapter 3. The MSP reported in [9] supports messaging features in the form of SOAP and REST style mobile web services, to be consumed by standardized SOAP and REST web service clients. MSP reported in [9] lacks cellular interfaces to handle IMS service clients and protocols to enable messaging services in the operator IMS network. This dissertation introduces standard IMS specific interfaces and protocols to the MSP and creates their bindings with the existing [9] server components and modules. Further, it is explained how prototypes of M2M text messaging and file sharing services are implemented to run over the operator's IMS network. The key features of the proposed messaging framework are:

- The text messaging service is realized in a request-response fashion together with the mapping of SIP MESSAGE mechanism over REST-synchronous architecture of the Mobile Server Platform.
- The file sharing service is realized in a session-oriented fashion together with the mapping of SIP INVITE mechanism over REST-asynchronous architecture of the Mobile Server Platform.
- The MSRP protocol is implemented to support session-oriented multimedia transfers in the IMS network.

The performance aspects of transport protocol and file transmissions are evaluated. It is noticed that the session establishment latency is independent of the file size and dependent on the underlying IMS network performance. The file transfer latency is dependent on the network speed and processing power of the mobile devices. The network speed is perceived to be stable and, therefore, the bottleneck to achieve good transfer rate is the processing capability of the mobile devices.

# Chapter 5

# Mobile-to-Mobile Multimedia Streaming Framework

This chapter presents the design and implementation of M2M multimedia streaming framework of the MSP. The implementation supports the standardized protocols and interfaces of the IMS network. The proposed framework is aimed to conform to the functional and non-functional implementation requirements for the RTP based audio and video streaming applications addressed by the 3GPP telecom association. These requirements are summarized in section 5.1. The chapter describes how M2M multimedia messaging streaming is realized in a service oriented fashion. Furthermore, the chapter provides the performance evaluation of the newly implemented streaming service prototype. The work presented in this chapter is mainly based on the author contribution in [22, 31]. Moreover, a master thesis guided by the author [13] has elaborated implementation details and conducted some additional measurements on the topic.

## 5.1 Multimedia Streaming Requirements in IMS/LTE network

3GPP specification TS 26.411 [4] for multimedia streaming applications define implementation requirements for RTP based audio and video application for an IMS client. Moreover, the 3GPP have provided a set of standardized requirements for the LTE and future mobile networks in order to support the QoS for multimedia application. In the scope of the dissertation, this section identifies and describes some key requirements in this regard:

***Interoperability*** The 3GPP requires all service providers and network operators to follow a common set of guidelines to enable multimedia content sharing or streaming for an IMS client. In this way, the interoperability of the applications between different networks can be made uniform.

***IP interface*** The IMS client should support an IP based interface for the transportation of media data and session control. The SIP protocol shall be used for the Control Plane signaling, whereas, RTP/UDP shall be used for media distribution.

***Session establishment*** The session setup procedure should be adopted by IMS clients, which leads to the idea that client should offer one RTP profile such as AVP (audio video profile) in the SDP offer message to the server.

***Interworking*** The later message interaction between the IMS client and the service provider server may contain more than one profile in order to support interworking.

***Video streaming parameters*** In case of video streaming the requirement during the session setup is to determine the RTP profile, bandwidth and video code, however, the "frame size" attribute in case of video streaming should not be used during the session setup.

***RTCP transmission*** Leading to the idea of AVP RTP profile in SDP message as per 3GPP specification, the IMS client can also implement the support of RTCP transmission in order to provision the Audio-Visual Profile with Feedback (AVPF).

***Sending and receiving ports*** An IMS client is to support the usage of same port number for sending and receiving RTP packet. However, to support interworking, the IMS client should be capable enough to receive packets on an alternate port.

***Packetization and segmentation*** The IMS client should support the packetization and segmentation whenever the RTP streaming communication is carried out between the two devices, therefore, in case of video streaming, the packetized data transfer during the session will ease the communication.

***3GP video format*** The 3GPP supports the usage of ISO (International Organization for Standardization) base 3GP video format for video services [release 5]. The

ISO introduced 3GP as a light weight file format for packet based networks (such as IMS) to offer video services. It's size is comparatively less than any other video file format, hence, it is the most recommended file format for portable devices. [51]

***QoS requirements*** Last but not least, 3GPP have provided a set of QoS requirements for the LTE and future mobile networks to support multimedia sharing and streaming applications. In LTE networks, the QoS is applied from the end user to the packet Gateway including a set of bearers. The dedicated bearer in LTE represents to a category where QoS needs to be provided for some specific applications, such as video streaming and VoIP, etc. Figure 5.1 depicts the QoS requirements for different multimedia applications in the LTE network. The bearer type Guaranteed Bit Rate (GBR) is used for the VoIP call, video call and video streaming applications, whereas, the Non-GBR supports the chat, ftp, email and IMS Signaling related applications.

| QCI | Bearer Type | Priority | Packet Delay | Packet Loss | Example |
|---|---|---|---|---|---|
| 1 | GBR | 2 | 100 ms | $10^{-2}$ | VoIP call |
| 2 | | 4 | 150 ms | $10^{-3}$ | Video call |
| 3 | | 3 | 50 ms | | Online Gaming (Real Time) |
| 4 | | 5 | 300 ms | $10^{-6}$ | Video streaming |
| 5 | Non-GBR | 1 | 100 ms | | IMS Signaling |
| 6 | | 6 | 300 ms | | Video, TCP based services e.g. email, chat, ftp etc |
| 7 | | 7 | 100 ms | $10^{-3}$ | Voice, Video, Interactive gaming |
| 8 | | 8 | 300 ms | $10^{-6}$ | Video, TCP based services e.g. email, chat, ftp etc |
| 9 | | 9 | | | |

FIGURE 5.1: QoS requirements of variuos mutlimedia application in LTE [24]

## 5.2 Realization of M2M Multimedia Streaming

This section explains how the existing MSP [9] is extended to realize M2M multimedia streaming over the operator IMS network. Like the other M2M services in the MSP, such as text messaging and multimedia file sharing, multimedia streaming is realized in a service oriented fashion. Therefore, a new service is implemented for this purpose, named as "MM Streaming".

The multimedia streaming service is implemented by using the asynchronous mechanism of MSP. Figure 5.2 depicts the call flow mechanism of multimedia streaming service over the operator IMS network. Similar to file sharing service (described in section 4.2.2), a SIP INVITE request from one MSP is sent to another MSP in order to create a multimedia session between two parties. The actual media transfer starts after the completion of session negotiations. In contrast to MSRP protocol in file sharing service, the multimedia streaming is achieved with 3GPP recommended RTP and RTSP protocols. The following section enfolds the message flow details of session establishment and media delivery phase.

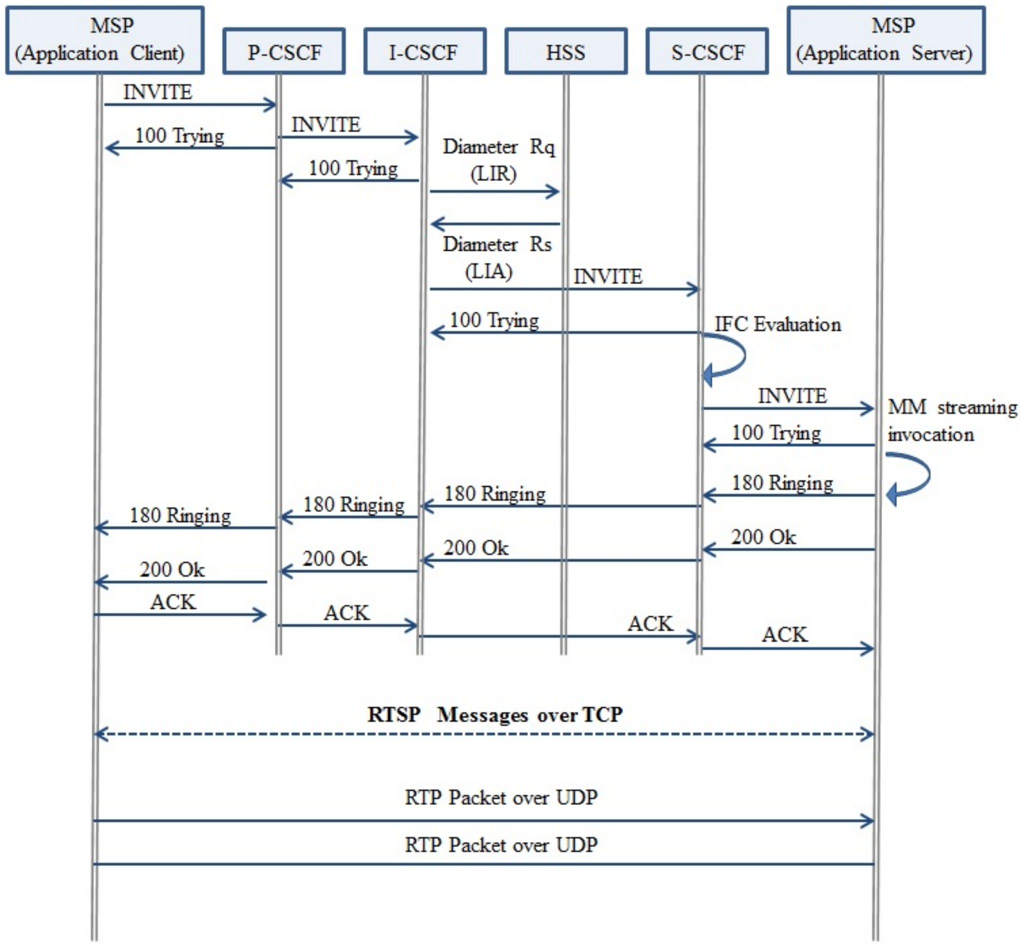



Figure 5.2: Call flow of M2M mutlimedia streaming service

## 5.3 Session Establishment Phase

SIP and SDP are the main protocols which are used for session negotiations in multimedia streaming service. In general, the steps involved in session negotiations for multimedia streaming service are exactly the same as described for file sharing service in section 4.2.2. However, the messaging constructs and attributes involved in SDP Offer-Answer model of multimedia streaming service are quite different from the file sharing service.

Listing 5.1 shows a sample SIP INVITE request which is sent from a user Bob to Alice in order to access the "MM Streaming" service. Here the SDP message content is as follows:

v - indicates the SDP version v=0

o - lists the organization of the calling party and the IP version is 4 with actual IP address 192.168.2.101

s - describes the SDP message as IMS Call

c - 192.168.2.101 is the IP address of the originator

m - describes the media type as RTP/AVP expected by the Bob. The port number (6022) is where Bob wants to receive the audio.

a - gives the media attributes PCMU, PCMA and telephone-event which are mapped with number 0, 8 and 11

The SDP message basically contains two type of information: session information and media information. In the SIP INVITE request, the lines starting with v,o,s,p,c,t indicate the session information i.e., version and user identifiers (v= and o= lines), subject of the session (s= line), phone number to obtain the information about the session (p = line), Bob's IP address (c= line) and time of the session (t = line). Whereas, the lines starting with m and a indicate media information i.e., media specific information (m = lines) and further information about the media (a = lines). The lines "a=curr : qos" and "a=des : qos" are related with QoS information that will be discussed later in section 6.1.3.

Once the SIP INVITE request is reached to the Alice's MobAS, a "100 Trying" message (listing 5.2) is sent back as an acknowledgment. Then Alice's MobAS reads the SDP content for the requested service name and the corresponding session and media related attributes. If everything is available and agreed, it sends a "180 Ringing" message (listing 5.3) back towards Bob to inform that the called side is being rung. Then Alice's MobAC answers the ringing with a "200 OK" (listing 5.4) message and wait for an acknowledgment (ACK message) from Bob. Once the "ACK" message (listing 5.5) is received, Alice's MobAS invokes the multimedia streaming service finally.

```
INVITE sip:alice@192.168.2.102:5070 SIP/2.0
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 INVITE
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" sip:alice@192.168.2.102:5070
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bK1a089c54d1c77dd448370e60e0bdf1e0323439
Max-Forwards: 70
Contact: "bob" sip:bob@open-ims.test:5070
Route: sip:192.168.2.105:6060;lr
My-Header: my header value
Content-Type: application/sdp
REST-Service-Uri: GET/Request-Response/Factory/createInstanceRq/
MMStreaming
resourcemethod: GET
Allow: INVITE, ACK, CANCEL, BYE, MESSAGE, OPTION, NOTIFY, PRAK,
UPDATE, REFER
User-Agent: UCT IMS Client
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
Privacy: none
P-Access-Network-Info: IEEE-802.11a
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel
Content-Length: 365
v=0
o=- 0 0 IN IP4 192.168.2.101
s=IMS Call
p=+46 8 52018010
c=IN IP4 192.168.2.101
t=0 0
m=audio 6022 RTP/AVP 0 8 101
b=AS:64
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=fmtp:101 0-11
a=curr:qos local none
a=curr:qos remote none
```

```
a=des:qos none local sendrecv
a=des:qos none remote sendrecv
```

Listing 5.1: SIP INVITE request for multimedia streaming service

```
Response received : Status Code = 100 CSeq: 20 INVITE
SIP/2.0 100 trying -- your call is important to us
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 INVITE
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" sip:alice@192.168.2.102:5070
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bK1a089c54d1c77dd448370e60e0bdf1e0323439
Content-Length: 0
```

Listing 5.2: SIP 100 Trying

```
Status Code = 180 CSeq: 20 INVITE SIP/2.0 180 Ringing
Record-Route: sip:mt@scscf.open-ims.test:6060;lr
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 INVITE
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" <sip:alice@192.168.2.102:5070>;tag=4321
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bK1a089c54d1c77dd448370e60e0bdf1e0323439
Contact: "alice" sip:192.168.2.102:5070
Content-Type: application/sdp ; Content-Length: 365
```

Listing 5.3: SIP 180 Ringing message

```
Status Code = 200 CSeq: 20 INVITE
Complete Response: SIP/2.0 200 OK
Record-Route: sip:mt@scscf.open-ims.test:6060;lr
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 INVITE
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" <sip:alice@192.168.2.102:5070>;tag=4321
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bK1a089c54d1c77dd448370e60e0bdf1e0323439
Contact: "alice" sip: 192.168.2.102:5070
```

Listing 5.4: SIP 200 OK

```
Sending ACK/nACK sip:192.168.2.102:5070 SIP/2.0
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 ACK
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bKaed3d158a9d993a2d729eff1406a454b323439
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" <sip:alice@192.168.2.102:5070>;tag=4321
Max-Forwards: 70
Contact: "bob" sip:bob@192.168.2.101:5070
Content-Length: 0
```

LISTING 5.5: SIP ACK message

## 5.4 Multimedia delivery Phase

RTP (Real Time Transport Protocol) [75], UDP(User Datagram Protocol) [75], RTSP (Real Time Streaming Protocol) [74]and RTCP (Real Time Control Protocol) [82] are some major protocols which are used in most of the multimedia streaming applications of Internet [55]. The RTSP is an application layer protocol that works with the lower layer protocols (e.g. RTP/UDP) to control seamless delivery of multimedia data over IP networks. The RTP provides support to UDP in real time data transportation. Whereas, the RTCP is used to send out-of-band control information for an RTP flow to provide feedback on the QoS being provided by the RTP.

As shown in figure 5.2, the RTSP signaling is used to acquire remote control functionality by creating a TCP connection between two communicating parties. Whereas, the multimedia data is transmitted as RTP packets (encapsulated inside UDP) from server to client. The following section explains how RTSP protocol works together with REST

### 5.4.1 RTSP mapping over REST Architecture

In the earlier version of MSP [9], some (six) major RTSP methods were proposed to be mapped over REST architecture: OPTIONS, DESCRIBE, SETUP, PLAY, PAUSE and TEARDOWN [31]. The OPTIONS, DESCRIBE and SETUP methods were used to establish multimedia sessions between two parties, whereas, the PLAY, PAUSE and TEARDOWN methods were used to provide control of streaming applications.

In the scope of this dissertation, the OPTIONS, DESCRIBE and SETUP methods are discarded and replaced with SIP and SDP protocols which are used to facilitate multimedia session negotiations in the operator IMS network, as explained in the previous section. Whereas, the remaining three methods, such as PLAY, PAUSE and TEARDOWN methods are utilized to control real time multimedia streaming in the IMS network. These methods are mapped over asynchronous REST architecture as shown in Table 5.1.

When a PLAY request is received by the multimedia streaming service, its state is changed from ready state to play. Similarly, when a PAUSE request is received by the service, the service comes to its earlier state i.e., the ready state. Finally, when a TEARDOWN request is sent by the client, the service is stopped and a SIP BYE message is sent to the client.

| RTSP Method | REST Style | RTSP States | Request Type |
|---|---|---|---|
| OPTIONS | NA | NA | NA |
| DESCRIBE | NA | NA | NA |
| SETUP | NA | NA | NA |
| PLAY | Asynchronous | rtsp.playing | changeState |
| PAUSE | Asynchronous | rtsp.ready | changeState |
| TEARDOWN | Asynchronous | rtsp.init | changeState |

TABLE 5.1: RTSP mapping over REST

### 5.4.2 RTP over UDP

The simplest optimized RTP frame structure is proposed to attain seamless deliverance of multimedia content in M2M service networks. Considering the Audio On Demand (AOD) model and assuming a single source audio, the figure 5.3 depicts an optimized RTP frame structure used for multimedia streaming over operator network.

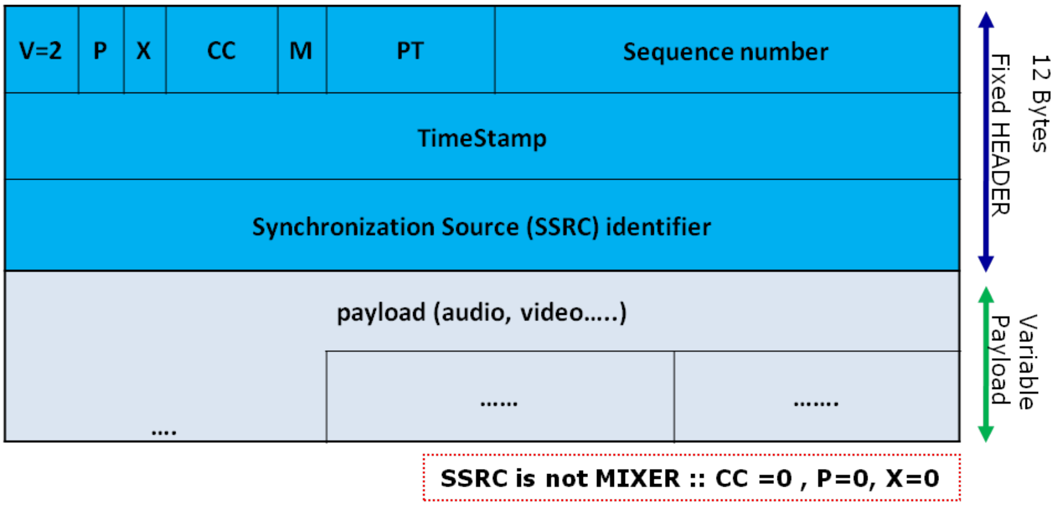



FIGURE 5.3: Implemented RTP Frame Structure [31]

Figure 5.3 depicts the 12 bytes fixed header and the least minimum requirement of RTP Header inside the RTP packets. In this RTP header, version (V) is 2. Padding (P) bit is set to “0” as there are no padding octets that are inserted by the application. The next Extension bit (X) is also set to “0” as there is no need of Extension header. The next Contributor Count (CC) value is also set to “0” and SSRC list is absent as there is only one single source and Mixer is not required here. The use of Marker (M) bit is based on application and in our case, it is used to identify the ending of the steaming file. So the last streaming RTP packet has this bit value “1” otherwise for the remaining packets, its value is “0”.

The Payload Type contains any dynamic value from 96-127 as our source data is mostly be of experimental format and does not come in the category of static payload types. Sequence number field has value starting with any random number (of 16 bits) in the first packet which is then linearly incremented in the following packets. The SSRC identifier contains any 32 bits random value that remains the same in all packets. For the Timestamp, it is any 32 bits random value in the first packet which is going to be incremented linearly in the following packets with the offset equal to Sampling Period.

When the RTP header fields are set inside the RTP packet, then the data bytes are read from the source media file and they are inserted as RTP payload inside the RTP packet. Finally this RTP packet is encapsulated inside the UDP datagram after which it is sent towards the client. The length of number of data bytes that can be inserted as RTP payload inside the RTP packet depends upon the Maximum Transmission Unit (MTU) allowed on the network path between the server and the client. For example, if the MTU is of 1500 bytes, then the maximum RTP payload size is `1500 - (20 + 12 + 8) = 1460` bytes.

## 5.5 Performance Evaluation

A prototype application was implemented in order to evaluate the performance of the newly created multimedia streaming service. The application allows the two network users to create an audio/video streaming session between them. As a usecase, a user Alice requests another user Bob to start a video streaming session. Here, the Service Creation Time (SCT) represents the time utilized in the construction of the requested service at Bob's MSP. This is a time between receiving the initial INVITE request and sending a 200 Ok response back. Whereas, the Service Invocation Time (SIT) represents the time required by the MSP to bring the demanded service in a ready state. This is a time taken by Bob's MSP between sending a 200 Ok response and starting a service thread after receiving an ACK from Alice.

Once the session is created, Bob sends a video URL to Alice over a TCP connection. This time is recorded as URL Receive Time (URT). After receiving the video URL, Alice's MSP starts a streaming connection (UDP) with the target server. Consequently, the Android VideoView API starts playing the video on the Alice's device. Here, the Video Streaming Time (VST) is calculated from starting the video at Alice's device till its end.

For evaluation purposes, the latencies such as SCT, SIT and URT are categorized as session establishment latencies. Whereas, VST will be represented as multimedia streaming latency.

### 5.5.1 Testbed

Performance evaluation by measure in the proposed framework has been extensively carried out in order to study the processing latencies during the session establishment and multimedia streaming phase. The testbed environment consists of two mobile user terminals and one server machine. The configuration of the testbed is as follows:

**User Terminal - 1 with MSP:**
Model: HTC Legend
Platform: Android 2.2
Camera: 5-megapixel color camera with auto focus and flash
Memory: ROM: 512MB RAM: 384MB
CPU Speed: 600MHz

**User Terminal -2 with MSP:**
Model: Samsung Galaxy S3 (i900)
Platform: Android 4.1.2 (Jelly Bean)
Camera: b-megapixel color camera with auto focus
Memory: ROM: 16 GB RAM: 1 GB
CPU Speed: Quad-core 1.4 GHz

**Server Machine with IMS Testbed:**
The Fraunhofer FOKUS OpenIMS Core is used as an IMS testbed. It comes as an Eclipse plug-in. It was running on the machine with the following specifications.
System Manufacturer: IBM
Model: Lenovo T41
Processor: Intel 1.7 GHz
Memory: RAM: 1 GB
Operating System: Linux Ubuntu 12.04
IDE: Eclipse JUNO

**Network Configurations**
ADSL: G.992.5 (ADSL2+)
Download: 17034 Kbps
Upload: 1149 Kbps
No of clients: 5

**Video Sources:**
Two video files were selected for the evaluation purpose:

**Video 1:**
Source: http://m.youtube.com/watch?client=mv-google
Length of video: 1:00 min
Name of video: Android in Space

**Video 2:**
Source: http://www.youtube.com/watch?v=QHIzzQB4Lew
Length of video: 0:45 second
Name of video: New Skype for Android

### 5.5.2 Session Establishment Latencies

Figures 5.4 and 5.5 depict the session establishment latencies of Video-1 and Video-2 files respectively. It is worthy to mention that the mean SCT value in Video-1 case is exactly the same to the corresponding mean SCT value in Video-2 case. Similar is the case of SIT i.e., the mean SIT value in Video-1 case is exactly the same to the corresponding mean SIT value in Video-2 case. The reasoning of same aggregated SCT and SIT values of both videos are because of the same processing time taken to complete service creation. So, for any video file of length (l), the SIT and SCT values will be constant depending on the processing speed of the device's processor. Here, one can notice that the given values of SCT and SIT are not exactly the same in figures. However, as the time measurements are taken in milliseconds, so this small difference is negligible.

On the contrary, the mean URT value in Video-1 case is smaller than that of the Video-2. However, this behavior of URT is unpredictable because its computation is strictly dependent on the network delay. If the network is good (small delay), the URL will be received in a short time. But, if the network is bad (big delays), the URL receive time will be bigger.

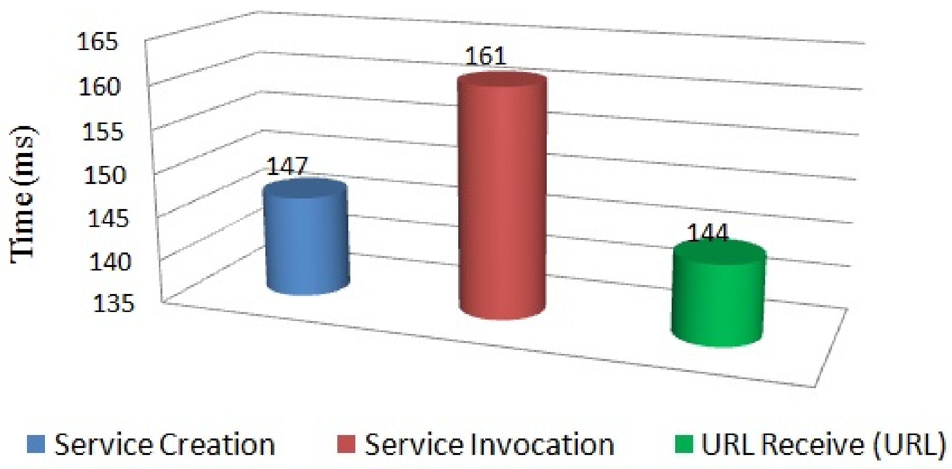



FIGURE 5.4: Video-1 Session establishment latencies

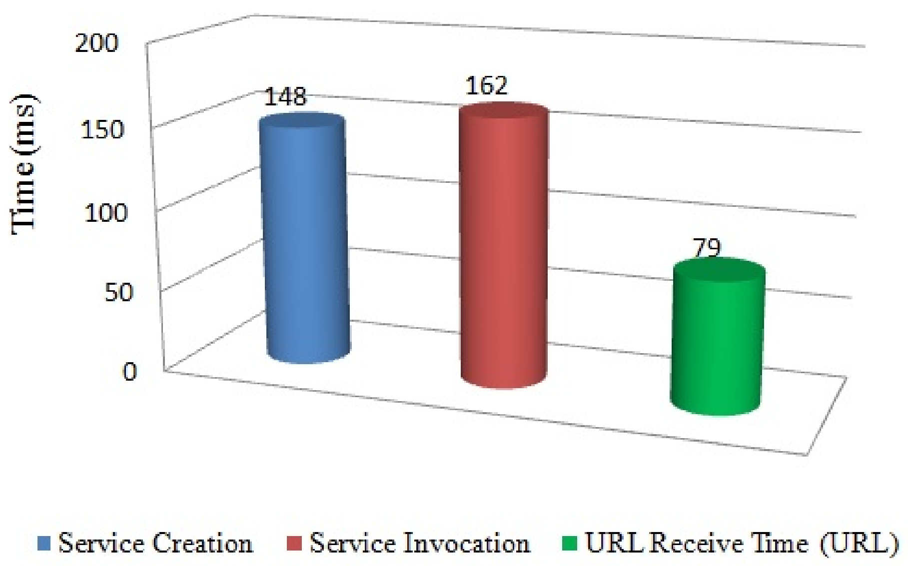



FIGURE 5.5: Video-2 Session establishment latencies

### 5.5.3 Multimedia Streaming Latencies

Figure 5.6 depicts the mean VST for a 45 seconds video (Video-1) is 48 seconds. Whereas the mean VST for a 60 seconds video is 64 seconds, as shown in figure 5.7. So by comparing the both results of Video-1 and Video-2, one can conclude that for every 15 seconds of the video, there is a 1 second delay produced by the network on average. However, a more detailed analysis on the multimedia streaming will be presented in section 6.3.

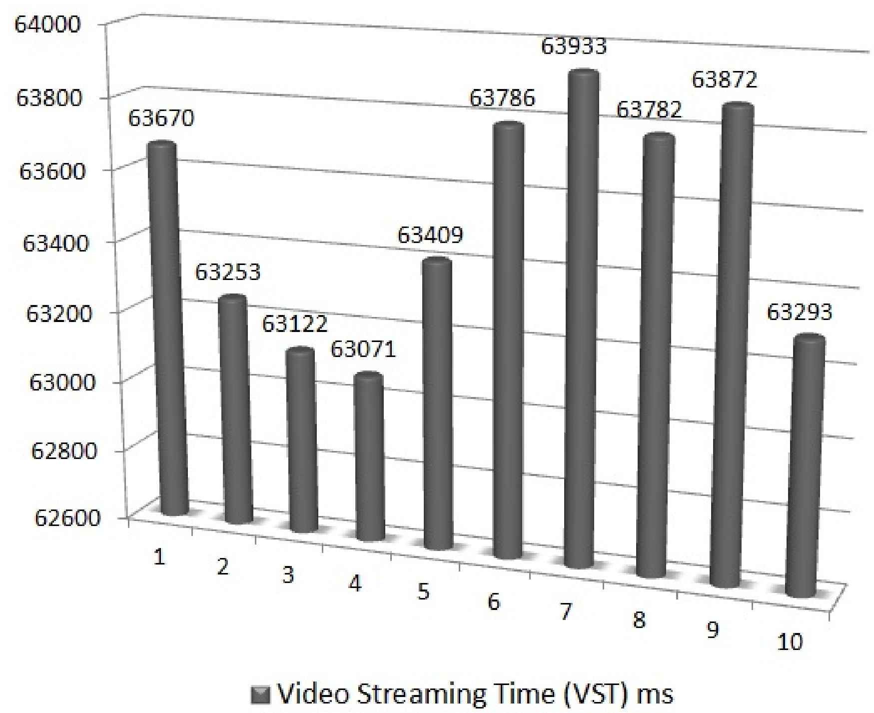



FIGURE 5.6: Video-1 video streaming times

## 5.6 Conclusion

This chapter presents the implementation and evaluation of M2M multimedia streaming framework for the IMS-Mobile Server Platform proposed in chapter 3. The MSP reported in [9] comprises a prototypical implementation of its M2M streaming

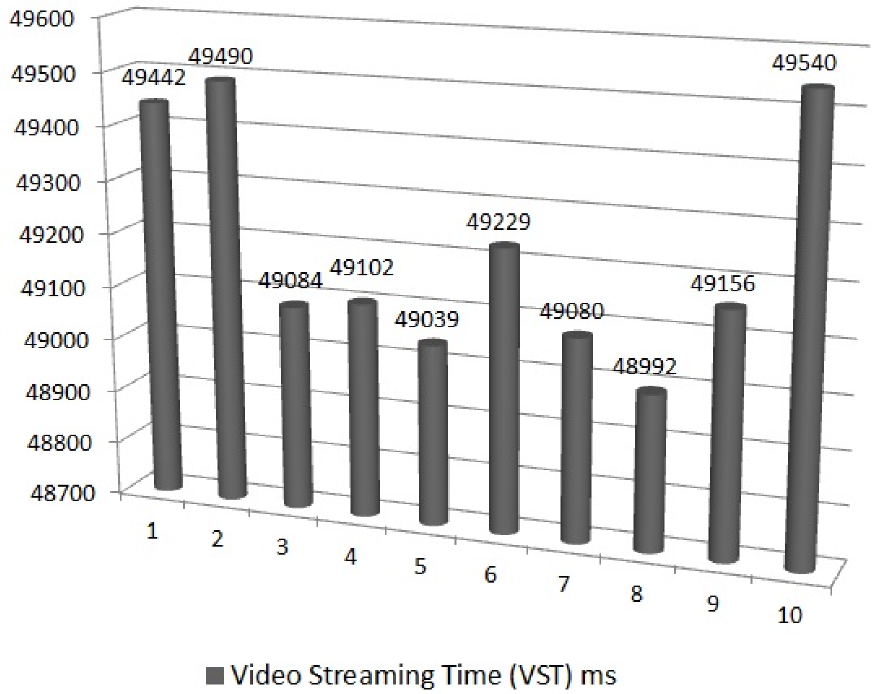



Figure 5.7: Video-2 video streaming times

framework that is limited to support standardized SOAP and REST interfaced service clients. Chapter 5 introduces the implementation of 3GPP standard conformant interfaces and protocols to realize M2M streaming over an operator IMS network. A prototype video streaming application is developed and tested as a proof of concept. The key features of the proposed multimedia streaming framework are:

- The multimedia streaming is realized in a session-oriented fashion together with the mapping of SIP INVITE mechanism over REST-asynchronous architecture of the Mobile Server Platform.
- RTSP protocol signaling is applied in order to achieve the remote control functionality in the multimedia applications, such as stopping, pausing and resuming the running media.
- A light version of RTP protocol is applied for multimedia transmission using UDP.

It is concluded that for any streaming video of length L, the service invocation time and service creation time will remain constant for a particular device processor. The video streaming time will be further discussed in detail in the following chapter.

# Chapter 6

# Mobile-to-Mobile Quality of Service Framework

The cellular world has witnessed a striking increase in mobile broadband subscribers and their traffic volume per subscriber. An obvious reason to this trend is the attractive data tariffs by operators e.g., flat-rate package and, on the other hand, the introduction of more advanced smart phone applications in the market. However, from the network perspective, provisioning high profile multimedia applications over the bandwidth-constrained cellular networks is not simply straight forward. QoS has always been an issue in this regard. Therefore, mobile operators have to employ more sophisticated control and policy enforcement mechanism in order to ensure the quality of individual services running over their network. More precisely, they require a precise way to do service differentiation and apply QoS policies accordingly.

This chapter presents the QoS framework of the MSP. The chapter enfolds how the proposed QoS framework operates in user registration, session establishment and multimedia delivery phase to provide an end-to-end QoS control for an existing multimedia streaming service. A promising feature of the proposed QoS framework is to allow the cellular operators to employ QoS control based on the user profiles classification. Hence, a systematic approach is adopted to study the behavior of a video streaming prototype based on different network settings and various user profiles.

The work presented in this chapter is mainly based on the author contribution in [22, 20]. Moreover, a master thesis guided by the author [13] has elaborated implementation details and conducted some additional measurements on the topic.

## 6.1 The life cycle of QoS Framework

This section defines how a complete end-to-end QoS control is achieved over the cellular operator network. The major stake holders of this life cycle are :

1) An end user, who generates and sends a session request to the operator network by using the client part of mobile server platform.

2) The operator network, which receives the requests from the end users and takes appropriate actions to ensure guaranteed QoS for a particular session.

3) A QoS evaluator module of mobile server platform which computes the latencies of a running session and evaluate quality of experience (QoE) accordingly.

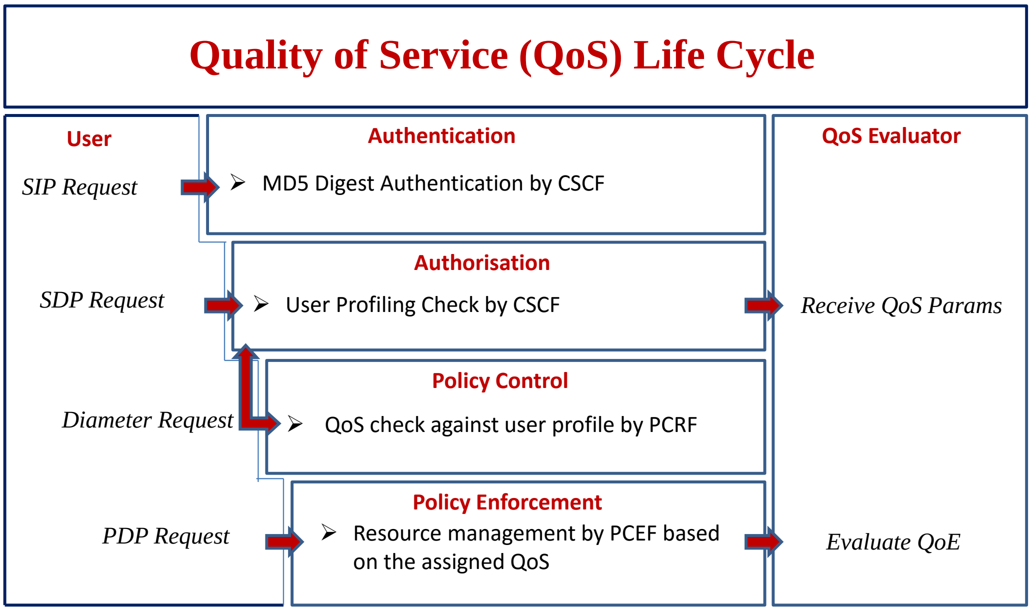



FIGURE 6.1: QoS life cycle

Figure 6.1 depicts how these stake holders play their active role in order to complete the QoS life cycle. The authentication and authorization process is a part-and-parcel of this life cycle. It ensures that only an authorized network user is able to invoke a service session in the network. Moreover, it ensures that the policy decisions such as user-specific limitations and general network policies are checked against the user requests and applied before an actual session starts. This process is called an early authorization. In an IMS network, it is the responsibility of CSCF functions to perform authentication and authorization of user profiles and sessions.

The policy control process is another integral part of the QoS life cycle. It takes control of the life cycle after the authentication and authorization process ends. Two types of policy control mechanisms come under the responsibility of this network process: charging control and QoS control. The charging control mechanism ensures that charging decisions are made and implemented against each user request. For instance, if a user runs out of credit the session should be terminated immediately. Whereas, the responsibility of QoS control mechanism is to read the QoS parameters requested by the user and check whether the network is able to provide such QoS guarantees for a particular session or not. For instance, if the network is not capable to provide more than a certain bandwidth during the peak hours, some heavy multimedia sessions requests shall not be entertained by the policy controller, which is known as PCRF (Policy and charging control function) in the 3GPP PCC architecture.

After the policy decisions are made by the policy controller, the assigned QoS parameters are shared with the PCEF (Policy and Charging enforcement Function), which plays a key role in the policy enforcement process of QoS life cycle. The decisions made by the policy controller are actually enforced by the PCEF in the real network. In an LTE-EPC system, the PCEF serves as a logical part of PGW (Packet Data Network Gateway) function, which acts as the interface between the LTE network and other packet data networks.

### 6.1.1 User Registration and QoS life cycle

Let us see QoS life cycle in operation at user registration phase. As explained in section 3.2.3.2, the MSP has to perform an IMS-level registration in order to access the operator IMS network. The registration process starts by sending a SIP register request to the IMS proxies and ends by receiving a final 200 OK response back. However, there are several other steps involved in this communication that can be seen in figure 6.2.

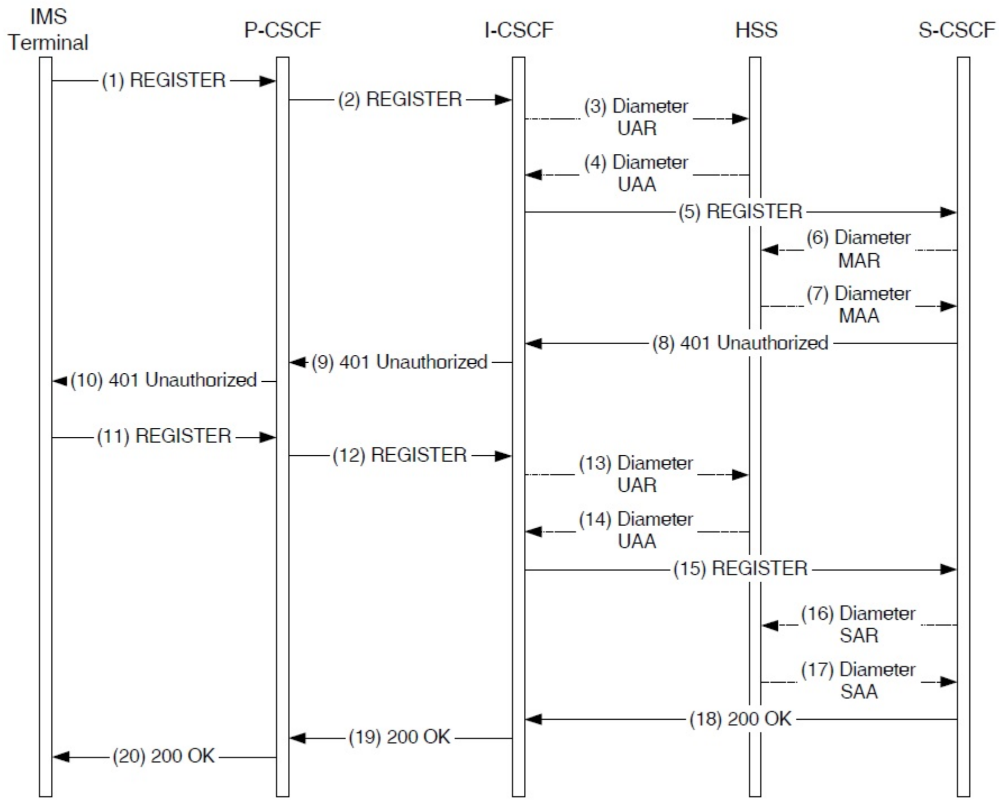



FIGURE 6.2: MSP Registration at the IMS level [28]

The important thing to note here is that when the initial SIP Register request is received by the S-CSCF, it has to authenticate the IMS user. There are several known authorization schemes that are supported by the IMS for this purpose. For example, a commonly used authorization scheme for SIP communication is HTTP Digest Access Authentication [47]. As per the rules defined in the standardized SIP specification [53], HTTP digest scheme must be supported by both client and server parties involved in the SIP conversation. Therefore, it is mandatory for a SIP server to authenticate a SIP client before the actual communication takes place. A SIP client can also authenticate a SIP server using this scheme but the decision is left upon the applications whether to implement this option or not. However, due to some security limitations, the HTTP Digest authentication scheme was not adopted by 3GPP in order to authenticate the cellular network users in the IMS network.

An advanced version of the HTTP Digest scheme, which is typically used for cellular networks, is called HTTP Digest Access Authentication using Authentication Key Agreement (HTTP Digest AKA) [67]. The basic aim behind the introduction of this scheme was to address the security issues of the earlier version. Hence, among other modifications, security improvements such as the mutual authentication and longer encryption keys were introduced in this scheme. In this scheme, a secret key used for authentication purpose is already stored in the smart card of the user devices, known as Universal Integrated Circuit Card (UICC). A major responsibility of UICC is to provide access to a certain cellular network and ensures the integrity and security of the user data. In order to access to a particular cellular network, the UICC contains one or several network applications collectively known as Network Access Applications (NAA). In GSM network, the application used for this purpose is called SIM (Subscriber Identity Module). Whereas for UMTS and LTE network, another application is used, which is named as USIM (UMTS Subscriber Identity Module). Although USIM may also be used to access the IMS network, there exists a specialized application for this purpose, known as ISIM (IP-Multimedia Services Identity Module). Each of these application stores a few configurations and parameters related to a particular usage.

#### 6.1.1.1 HTTP Digest Authentication Mechanism

As explained above, HTTP Digest Authentication is the simplest form of an authentication scheme used for SIP communication. It is based on a simple challenge-response paradigm and comprised of two registration cycles by the network user. In the first registration cycle, the user sends an initial SIP Register request to the IMS network and receives an authorization challenge response from the S-CSCF. In the second registration cycle, the user computes the response of the challenge message, embeds it to the initial register request and sends a re-Register request to the IMS network. The S-CSCF verifies the response and authorizes the user by sending a 200 Ok response finally.

The MSP implements this scheme to authenticate the users in the IMS network. Listing 6.1 depicts the initial SIP Register request which is sent by the MSP to the IMS network. Two headers are important to note in this request: CSeq header and Authorization header. The value of CSeq header, "1 REGISTER", confirms that this is the initial register request to the IMS network, which means the user is not earlier authorized by the IMS network and an authorization process must be invoked by the S-CSCF in response. Therefore as per the rules of Digest Authorization scheme, the values of the nonce and the response fields in the Authorization header are left empty in the initial register request. These values will be computed and sent in the second register request against the authorization challenge message by the S-CSCF.

```
REGISTER sip:bob@open-ims.test SIP/2.0
Call-ID: fa60d272af3e8d7ef33af7445c99322b@192.168.2.101
CSeq: 1 REGISTER
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "bob" sip:bob@open-ims.test
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bKb6f36916c7b7c863e069f72d6326905a323535
Max-Forwards: 70
Contact: "bob" sip:bob@192.168.2.101:5070
My-Header: my header value
My-Other-Header: my new header value
Call-Info: http://www.antd.nist.gov
Authorization: Digest username="bob@open-ims.test",
realm="open-ims.test",nonce=" ",uri="sip:open-ims.test",response=" "
User-Agent: UCT IMS Client
Expires: 600000
Supported: path,gruu
Content-Length: 0
```

Listing 6.1: Initial SIP Register Request

After receiving the initial SIP Register request, the S-CSCF sends an MAR(Multimedia-Auth-Request) Diameter request to the HSS in order to obtain the authentication vectors against the user. Based on these authentication vectors, the S-CSCF prepares an authorization challenge response and sends towards the MSP. Listing 6.2 shows the 401 unauthorized challenge message of S-CSCF, which contains an authentication header (WWW-Authenticate) with realm, nonce and algorithm values. It is important to note that the same Call-ID is used by the S-CSCF as in the initial SIP Register request, which reflects that the communication thread is the same as initiated by the user and the challenge message is the response of the register request.

```
Status Code = 401 CSeq: 1 REGISTER
Complete Response: SIP/2.0 401 Unauthorized - Challenging the UE
```

```
Call-ID: fa60d272af3e8d7ef33af7445c99322b@192.168.2.101
CSeq: 1 REGISTER
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "bob" <sip:bob@open-ims.test>;
tag=3fb7d972dbe65e87dd84f768b8532c2a-a4c6
Via: SIP/2.0/UDP 192.168.2.101:5070;rport=5070;
branch=z9hG4bKb6f36916c7b7c863e069f72d6326905a323535
Path: sip:term@pcscf.open-ims.test:4060;lr
Service-Route: sip:orig@scscf.open-ims.test:6060;lr
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,REFER,SUBSCRIBE,NOTIFY,
PUBLISH,MESSAGE,INFO
Server: Sip EXpress router (2.1.0-dev1 OpenIMSCore (i386/linux))
Warning: 392 192.168.2.105:6060 "Noisy feedback tells:
pid=25267 req_src_ip=192.168.2.105 req_src_port=5060
in_uri=sip:scscf.open-ims.test:6060
out_uri=sip:scscf.open-ims.test:6060 via_cnt==3"
WWW-Authenticate: Digest realm="open-ims.test",
nonce="6c0d2f4ad44175dbb79d1bd39f9f7917",
algorithm=MD5,qop="auth,auth-int"
Content-Length: 0
```

Listing 6.2: 401 - Unauthorized Challenge Response

Now the MSP extracts the values of authentication header from the 401 challenge message and utilizes the HTTP digest algorithm (Ref RFC 2069) in order to compute the authentication response. The implementation of HTTP digest algorithm is shown in the listing 6.3.

```
// Generate the response digest according to RFC 2069, i.e.
// response-digest = <"> < KD ( H(A1)":" unquoted nonce-value ":" H(A2
    ) ) > <">

// A1 = unquoted username-value ":" unquoted realm-value ":" user-
    password
A1 = registeredUser+"@"+"open-ims.test" + ":" + realm1 + ":" +
    password;

// A2 = Method ":" digest-uri-value
String A2 = method.toUpperCase() + ":" + "sip:open-ims.test";

// H(A1) = The digested value of A1 converted to a hex string
// H(A2) = The digested value of A2 converted to a hex string
```

```
String KD = H(A1) ":" nonce ":" H(A2)
```

Listing 6.3: HTTP Digest Algorithm

It is clear from the algorithm that the new response value is computed based on the nonce value of the challenge message. Therefore, the newly computed authorization header is embedded in the initial SIP register request and a re-register request is sent to the IMS network, as shown in figure 6.2. Listing 6.4 shows a comparison between the Authorization header of the initial SIP message and the newly computed reply/response Authorization header after the 401 unauthorized challenge messages.

```
REGISTER sip:open-ims.test SIP/2.0
Call-ID: fa60d272af3e8d7ef33af7445c99322b@192.168.2.101
CSeq: 2 REGISTER
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "bob" sip:bob@open-ims.test
Via: SIP/2.0/UDP 192.168.2.101:5070
Max-Forwards: 70
Content-Length: 0
Authentication message as header extension Authorization: Digest
username="bob@open-ims.test",realm="open-ims.test",
nonce="6c0d2f4ad44175dbb79d1bd39f9f7917",
uri="sip:open-ims.test",
response="78c5ba8640973ebecfeea180f3bd5be4",algorithm=MD5
User Agent Header User-Agent: UCT IMS Client
Expires: 600000
```

Listing 6.4: SIP Re-Register Request

Once the re-register request is received, the S-CSCF marks the Call-ID and the values of nonce and response in the Authorization header. After verifying the new values, it sends a 200 OK - SAR successful message back to the MSP and the user is registered with the IMS framework for further communication.

### 6.1.2 Session Establishment and QoS Life Cycle

After a successful registration in the IMS network, the MSP establishes a service session every time in order to start a new multimedia communication. The session is initiated by sending a SIP INVITE message from one MSP user to another. As explained in section 4.2.2.1 , an SDP offer is also sent inside the INVITE message which includes all supported codecs required to make a multimedia call between the two parties.

As shown in figure 6.3, the INVITE message form the originating MSP is received by the P-CSCF in the terminating IMS network. Before forwarding the INVITE message to the terminating MSP, the terminating P-CSCF first reads the media related information from the SDP offer and then requests a media authorization token (MAT) from the policy controller (PCRF) in order to check whether the terminating party is authorized to the requested media types and codecs. If the P-CSCF determines that a certain media type or codec should not be used in this session, an immediate rejection will be sent to the MSP by the P-CSCF. Otherwise, the INVITE message is sent to the terminating MSP. The terminating MSP receives this MAT in the INVITE message which needs to be passed to the network gateway (PCEF/PGW) later for resource reservation purpose.

The MSP prepares a common list of media codecs from the SDP offer and sends it back via a 183 progress message. Once the 183 progress message reaches the originating IMS network, the originating P-CSCF repeats the media authorization process and a MAT is sent to the originating MSP for resource reservation purpose.

### 6.1.3 QoS Differentiation Based on User Profiles and Service Types

Listing 6.5 shows a sample INVITE message with a complete SDP media description and QoS information embedded in the SIP body. The QoS pre-conditions, such as the lines a=curr:qos lines, indicate the actual and the current situation of the end parties that whether the resources needed for meeting the quality of service requirements are available or not. Whereas, the a=des:qos lines indicate the desired quality of service requirements needed for a particular multimedia session. Here, the lines "a=curr:qos local none" and "a=curr:qos remote none", indicate that there is no QoS terms and conditions determined between two parties at the moment.

Additionally, the dissertation proposes two new QoS related headers in the body of the SIP message: QoS-class-user and QoS-class-service. The motivation behind the introduction of these headers is to let the operators control the network traffic based on the different user profiles and service types. The QoS-class-user header specifies the users priority level based on their subscription data i.e., for prototype and evaluation purposes, the dissertation defines three priority levels based on the three different user profiles such as Gold, Bronze and Sliver user profile. Whereas, the QoS-class-service header specifies the service priority level based on different service types i.e., three priority levels are defined for three different services such as text messaging, multimedia file sharing and multimedia streaming. Some more details on these priority levels will be given in the performance evaluation section.

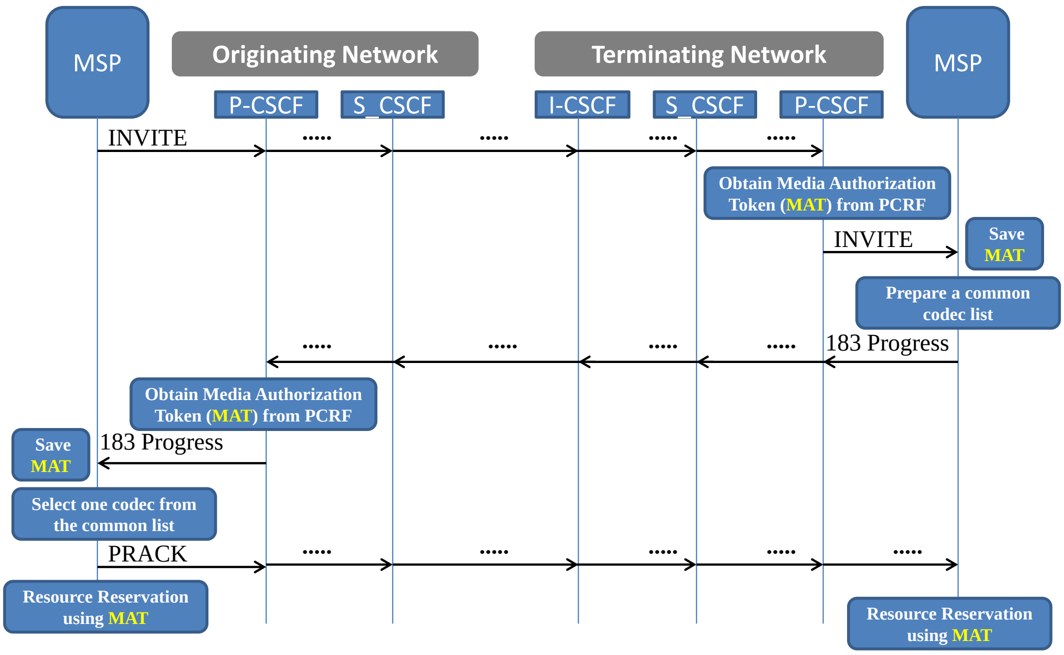



FIGURE 6.3: QoS control in Session Establishment

```
INVITE sip:alice@192.168.2.102:5070 SIP/2.0
Call-ID: d425d06c866342cdf29397226cf7ae14@192.168.2.101
CSeq: 20 INVITE
From: "bob" <sip:bob@open-ims.test>;tag=12345
To: "alice" sip:alice@192.168.2.102:5070
Via: SIP/2.0/UDP 192.168.2.101:5070;
branch=z9hG4bK1a089c54d1c77dd448370e60e0bdf1e0323439
Max-Forwards: 70
Contact: "bob" sip:bob@open-ims.test:5070
Route: sip:192.168.2.105:6060;lr
Content-Type: application/sdp
```

```
REST-Service-Uri: GET/Request-Response/Factory/
createInstanceRq/SStreaming
resourcemethod: GET
QoS-Class-user: 2
QoS-Class-service:0
Content-Length: 365
v=0
o=-0 0 IN IP4 192.168.2.101
c=IN IP4 192.168.2.101
m=audio 6022 RTP/AVP 0 8 101
b=AS:64
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=fmtp:101 0-11
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
```

Listing 6.5: QoS headers in INVITE Request

## 6.2 Media Delivery and QoS Life Cycle

Referring again to the figure 6.3, the originating MSP sends a PRACK message back to the terminating MSP in response to the 183 progress message. The PRACK is the last message before the MSP starts resource reservation for media delivery phase. The protocol used for resource reservation is called RSVP (Resource Reservation protocol) [26]. The MSP sends a RSVP request to the PCEF including the media authorization token (MAT). This is the same authorization token that was sent by the P-CSCF to the MSP during the session establishment phase. The PCEF forwards the token to the policy controller (PCRF) together with the requested QoS parameters. The PCRF then utilizes the authorization token to counter check that whether the requested QoS parameters match with the already negotiated QoS parameters in the session establishment phase.

This is the final authorization check performed by the network to allocate the resources for the upcoming media delivery session. Finally, the PCRF takes the policy decision and informs the PCEF that the MSP is allowed to use the allocated QoS resources. The PCEF then enforces the policy decision and allows the MSP to start multimedia traffic.

## 6.3 Evaluations

With the inclusion of QoS framework in the MSP, an extensive study is carried out to analyze its impact on the individual services. A systematic approach is adopted in order to study the session establishment and media delivery latencies for different user profiles and service types. For evaluation purpose, three different user profiles are created in the MSP:

**Gold user profile:** Gold user profile reflects to the user category who receive best available network conditions throughout the service session.

**Silver user profile:** Bronze user profile point to the user category who experience some network disruption (packet delay, Jitter, etc) during the service session.

**Bronze user profile**: This profile illustrates the user category who experience major network disruption and traffic irregularities during the service session.

### 6.3.1 Testbed

The testbed setup is consisted of two user terminals and one server machine acting as an IMS server. The configuration of the testbed is as follows:

**User Terminal - 1 with MSP:**
Model: HTC Legend
Platform: Android 2.2
Camera: 5-megapixel color camera with auto focus and flash
Memory: ROM: 512MB RAM: 384MB
CPU Speed: 600MHz

**User Terminal - 2 with MSP:**
Model: Samsung Galaxy S3 (i900)
Platform: Android 4.1.2 (Jelly Bean)
Camera: 6-megapixel color camera with auto focus
Memory: ROM: 16 GB RAM: 1 GB
CPU Speed: Quad-core 1.4 GHz

**Server Machine with IMS Testbed:**
Fraunhofer FOKUS OpenIMS Core is used as an IMS testbed. It comes as an Eclipse plug-in. It was running on the machine with the following specifications.
System Manufacturer: IBM
Model: Lenovo T41
Processor: Intel 1.7 GHz
Memory: RAM: 1 GB
Operating System: Linux Ubuntu 12.04
IDE: Eclipse JUNO

**Video Sources:**
Two video files were selected for the evaluation purpose:

**Video 1:**
Source: http://m.youtube.com/watch?client=mv-google
Length of video: 1:00 min
Name of video: Android in Space

**Video 2:**
Source: http://www.youtube.com/watch?v=QHIzzQB4Lew
Length of video: 0:45 second
Name of video: New Skype for Android

**File Sources:**
Two image files were selected for the evaluation purpose:

**File 1:**
Source: Own Camera Picture

Length of file: 7.02 Kb
Name of file: example1.jpg

**File 2:**
Source: Own Camera Picture
Length of file: 32 Kb
Name of video: example2.jpg

### 6.3.2 File Sharing Evaluation

As mentioned in section 4.3.2, file transfer time (FTT) is evaluated and analyzed for two different files. It is the time to complete the file transfer from one MSP to another, which does not include the session establishment time. Table 6.1 presents how file sharing experiments are performed in order to study the FTT behavior for different user profiles.

As indicated in table 6.1, the FTT is studied in two network conditions i.e., Fixed delay and Jitter. The motivation behind is to study the impact of different network conditions for different user profiles. However, the corresponding network delays are produced by the software simulations. The listing 6.6 presents a code snippet to produce random delay in case of Jitter condition.

```

/@ This method is used to generate random delay value within the range
    of Min Delay - Max Delay
private long delay_random() {
long delayRandom = delayMin + (int)(Math.random()*
((delayMax-delayMin) + 1));
randomDelayList.add(delayRandom);
return delayRandom;
}
```

LISTING 6.6: Code snippet to produce random delays

| User Profile | Fixed Delay | Jitter |
|---|---|---|
| Gold User | Delay = 0 | Min Delay = 0 Max Delay = 0 |
| Silver User | Delay = 250 ms | Min Delay = 200 ms Max Delay = 300 ms |
| Bronze User | Delay = 550 ms | Min Delay = 500 ms Max Delay = 600 ms |

TABLE 6.1: File Sharing Experiments

#### 6.3.2.1 File-1 FTT Evaluation

Figures 6.4, 6.5 and 6.6 depict the measurement results of FTT for Gold, Silver and Bronze user profiles respectively. In case of Gold user profile, the mean FTT in Jitter condition is slightly greater than that of Fixed delay condition. As in principle, the Gold user profile gets no ('0') network delay in both network conditions, so the difference here in FTTs is assumed as software delay i.e., the computation of random generator function after every 3 seconds in Jitter condition.

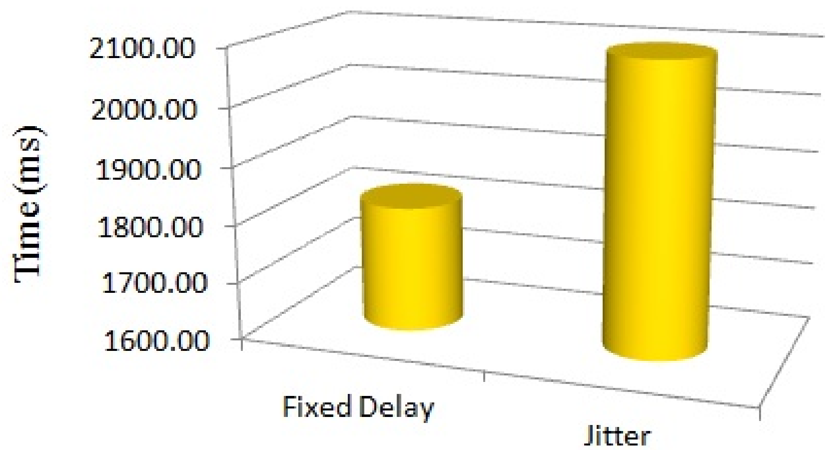



FIGURE 6.4: File 1 - FTT results for Gold users

In contrast to Gold user profile, the FTT behavior is somewhat opposite for Silver and Bronze user profiles. Figures 6.5 and 6.6 show that the mean FTT in Jitter condition is smaller than the one in Fixed delay condition. This FTT behavior actually validates the theoretical phenomena that network Jitter is not always worse than fixed delay condition. For instance, the Jitter value for Silver user is randomly generated between 200 and 300 ms (milliseconds), whereas the fixed delay value is 250 ms. So, there is a possibility that most of the times the random function generate a Jitter value lesser than 250 ms. In that case, the mean FTT in Jitter

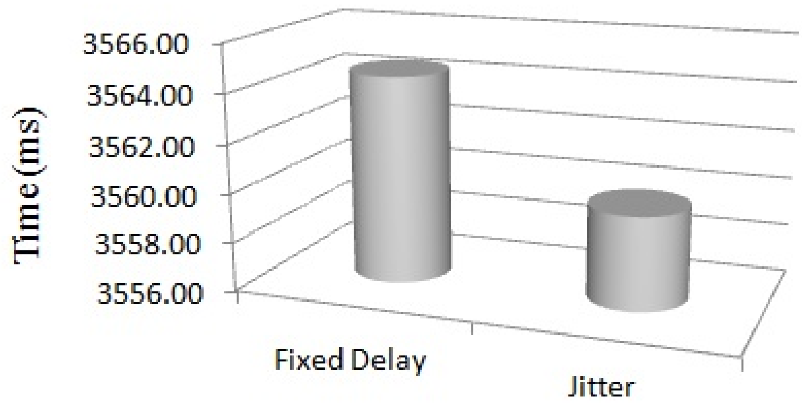



Figure 6.5: File 1 - FTT results for Silver users

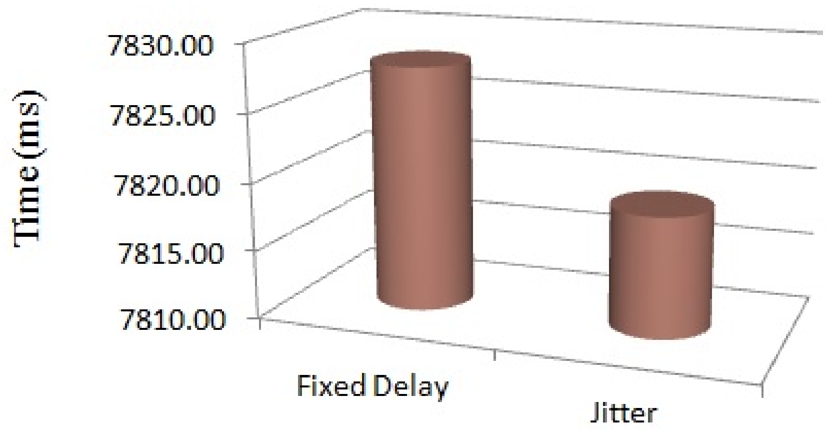



Figure 6.6: File 1 - FTT results for Bronze users

condition will be smaller than the one in Fixed delay. The similar scenario is true for Bronze user profile. However, this FTT behavior does not reflect the true picture of worst case scenario, where the Jitter delay may always be greater than the Fixed network delay. Such scenario will produce FTT results opposite to what are shown in figures 6.5 and 6.6.

#### 6.3.2.2 File-1 FTT Evaluation Based on User Profiles

Figure 6.7 presents a comparison of FTT measurements for different user profiles in the Fixed delay network condition. It is noticeable that the mean transfer time for Silver users is almost double than that of Gold users. Similarly, the mean transfer time for Bronze users is almost double than that of Silver users. So, for example, if a given file "F" takes "X" amount of transfer time for a Gold user, the same file will take 2X time for a Silver user and hence, 4X time for the Bronze user. From network operator's perspective, this relationship between different user profiles is important to predict and guarantee a certain QoS level for a given session.

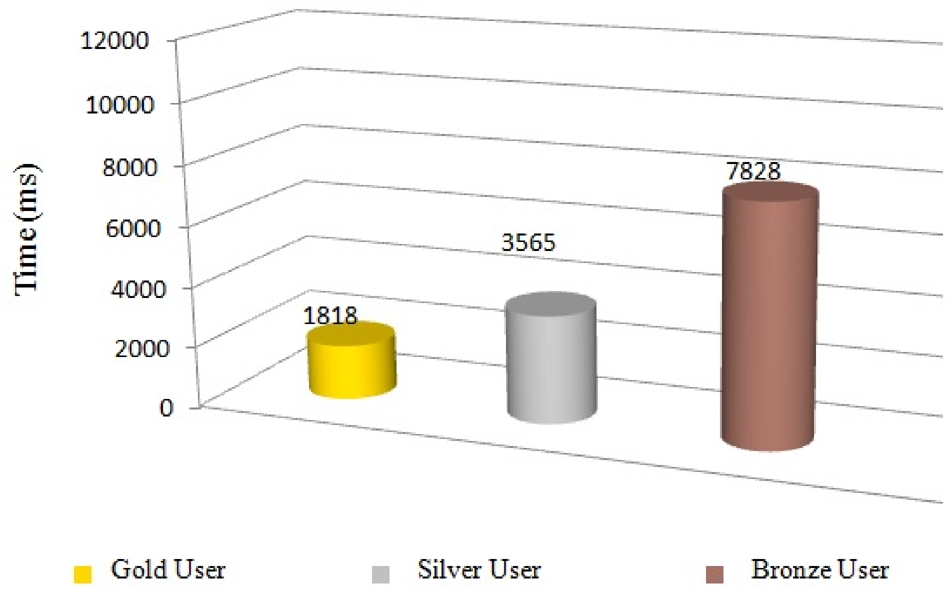



FIGURE 6.7: File 1 - FTT results in fixed delay network condition

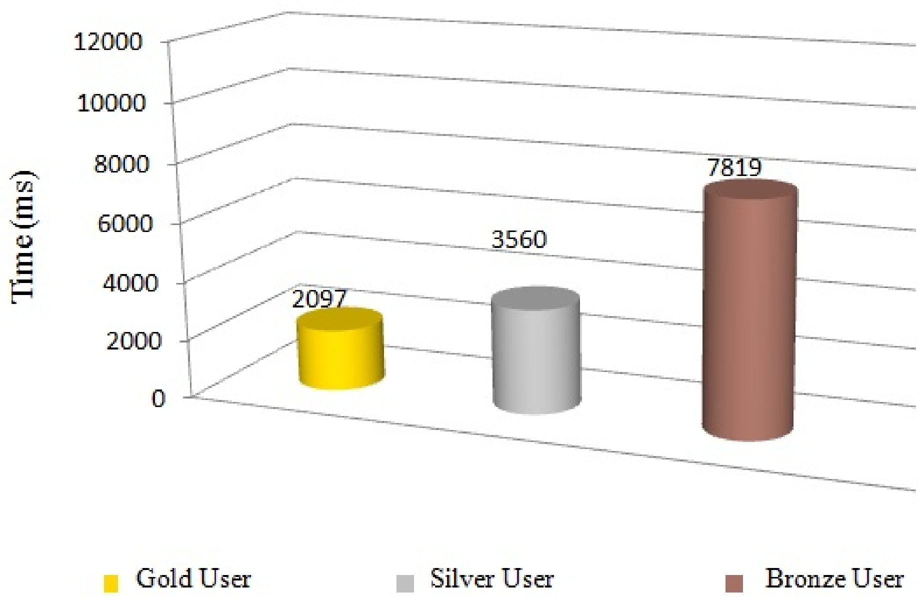



FIGURE 6.8: File 1 - FTT results in Jitter network condition

The similar FTT relationship can be seen for the given user profiles in case of Jitter network condition, as shown in figure 6.8. Hence, the mean transfer time for Silver users in Jitter condition is almost double than that of the Gold users. And, the mean transfer time for Bronze users is almost double than that of the Silver users. Here, one can identify that the FTT measurements shown in the figure are not exactly double to each other for different user profiles. However, as the time measurements are taken in milliseconds, so this difference is negligible. Finally, it is worthy to mention that this relationship holds only for this particular case. In order to generalize such relationship, there are more experimental evidences required with some other files of different sizes.

#### 6.3.2.3 File-2 FTT Evaluation

The same experiments were performed to study the FTT behavior of another file with same network settings and conditions. Figures 6.9, 6.10 and 6.11 present the results for Gold, Silver and Bronze user profiles in this regard. Here, the FTT behavior for Gold and Bronze user profiles is similar to that of File 1. Hence, for Gold users, the file transfer in Jitter condition takes more time than in Fixed delay condition, whereas, for Bronze users, it takes less time in Jitter condition than that of Fixed delay condition.

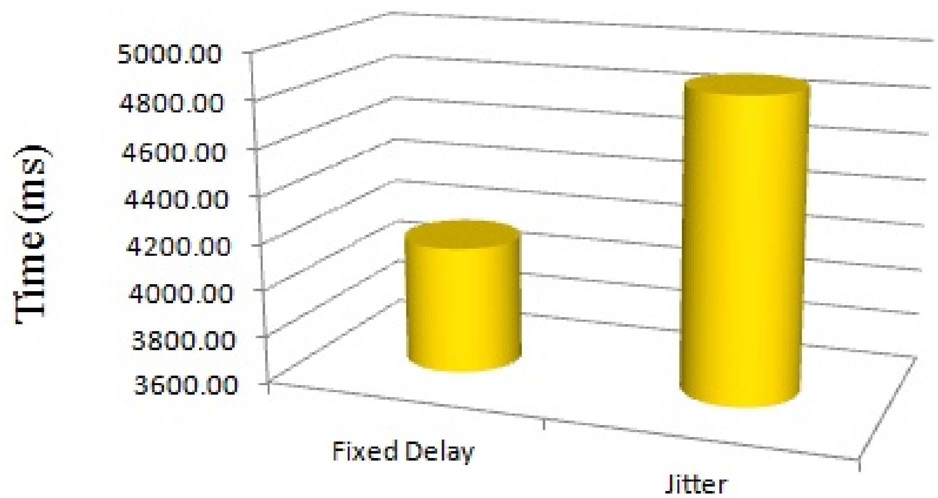



Figure 6.9: File 2 - FTT results for Gold users

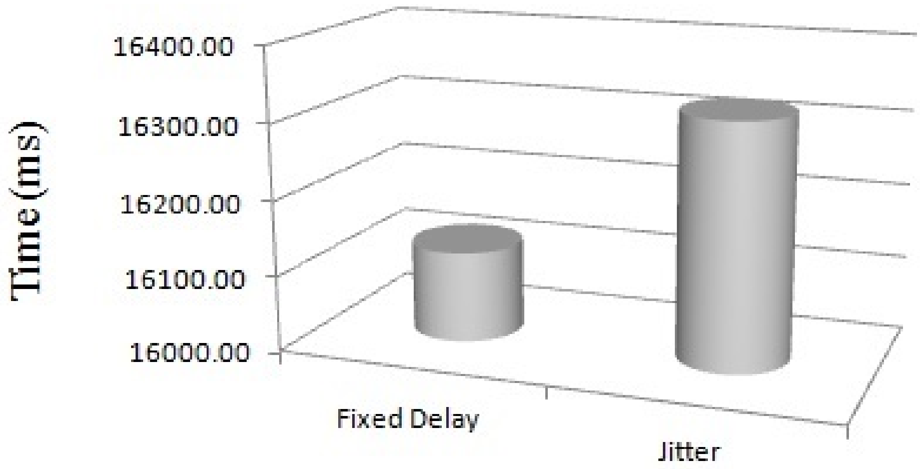



Figure 6.10: File 2 - FTT results for Silver users

However, the FTT results for Silver user profile are opposite to that of File 1. Hence, in contrast to File 1, the file transfer takes more time in Jitter condition than in Fixed Delay network condition. As explained earlier in section 6.3.4.1, this case reflects to the worst case scenario i.e., the network delay produced by Jitter is greater than the Fixed network delay.

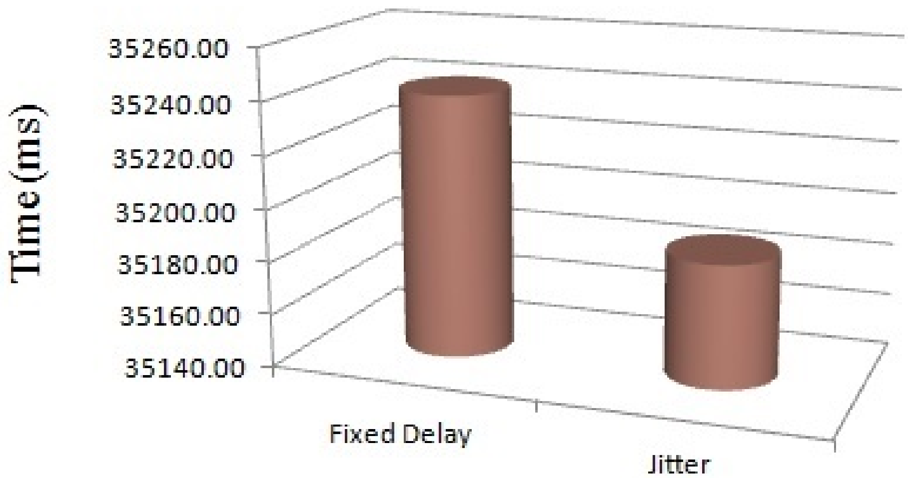



Figure 6.11: File 2 - FTT results for Bronze users

#### 6.3.2.4 File-2 FTT Evaluation Based on User Profiles

As in section 6.3.2.2 for File 1, figures 6.12 and 6.13 show the comparison of FTT measurements for different user profiles for File 2. Here, the FTT results obtained for File 2 donot fully acknowledge the earlier finding of mathematical relationship between the user profiles i.e., the mean transfer time for Silver users in Jitter condition is double than that of the Gold users, and, the mean transfer time for Bronze users is double than that of the Silver users. Moreover, one can notice that there exists still some relationship among the obtained FTT results i.e., the mean transfer time for Silver users in Jitter condition is four times greater than that of the Gold users, and, the mean transfer time for Bronze users is somehow double than that of the Silver users. Again, the measurements shown in the figure are not exactly double or four times to each other for different user profiles but the difference is negligible.

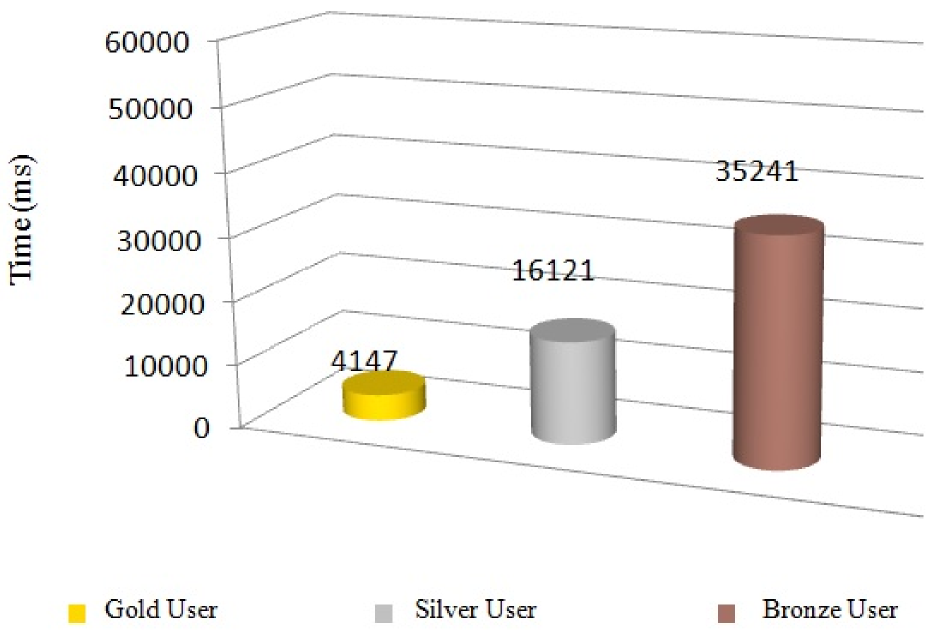



FIGURE 6.12: File 2 - FTT results in fixed delay network condition

### 6.3.3 FTT Mathematical Modeling

Two modes of evaluations for file sharing application are shown in the above section:

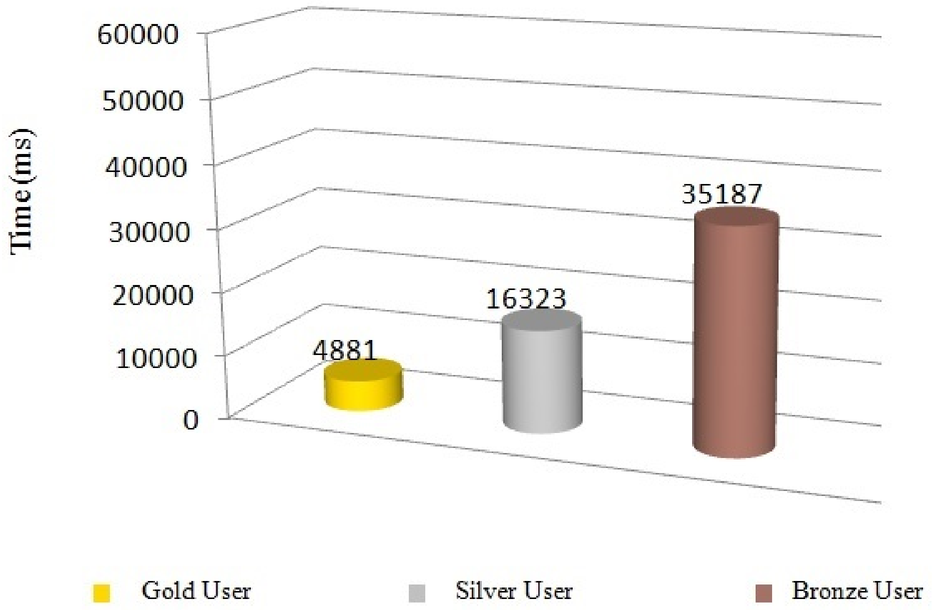



FIGURE 6.13: File 2 - FTT results in Jitter network condition

**FTT Comparison:** FTT evaluations for two different files in order to study the comparison of FTT behavior for three different user profiles for a given network condition.
**FTT Behavior:** FTT evaluations for two different files in order to study the FTT behavior of each individual user profile based on the given network condition i.e., Fixed packet delay and Jitter.

In the subsection below, the mathematical models are presented for the said modes of FTT evaluation. The overall objective is to validate the obtained experimental results in the above section and to propose a sophisticated mathematical tool that may help the network operators to predict the FTT behavior for different network settings and for various user profiles.

#### 6.3.3.1 FTT Comparison Based on User Profiles

Let us assume the Gold user profile as a reference model for our FTT evaluations i.e., no network delay and network variations for a given FTT session. The rest of the user profile models (such as Silver and Bronze profiles) can be easily derived with respect to the reference model. In the simulation setup used as a testbed for underlying network, the following network settings are modeled for Fixed packet delay case:

**Initial Delay:** An initial delay is induced at the beginning of each FTT session.
**Fixed Interval Delay:** A fixed length packet delay is induced at every 3 seconds of an ongoing FTT session.

Therefore, for a given FTT reference model the FTT(S) for Silver user profile can be derived as:
$FTT(S) = FTT(G) + DLi + DL(S) * Ni + DSi + Ds * Ni + Vnet$
where, $FTT(G)$ is FFT computed for Gold user profile,
$DLi$ is the initial delay induced in the network,
$DL(S)$ is the fixed interval delay,
$DSi$ is the software delay induced by the initial delay generation function,
$Ds$ is the software delay induced by the fixed delay generation function,
$Vnet$ is the delay produced by the network variations or fluctuations for an ongoing session,
and $Ni$ is the number of iterations how many times a fixed delay is produed for a complete FTT span.

In the simulation setup, the length of the initial delay is kept equal to the fixed interval delays for simplicity i.e., DLi = DL(S) and DSi = Ds. So, simplifying the above equation:
$FTT(S) = FTT(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) + Vnet$
and

$$FTT(S) = FTT(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) \tag{6.1}$$

where, $Vnet$ can be assumed as '0' if a dedicated path (with no fluctuations) is reserved for a particular service session.

The only thing which is unknown in the above equation is the Ni. However, as we know that the delay function is called after every 3 seconds, so the value of Ni can be easily found with the following formula.

$Ni = FTT(G) + DL(S) + Ds + Vnet/3000$
where $Ni$ is an integer i.e., the division sign is for integer division (non-fraction). It is worthy to note that Ni is dependent on the network variations. That means, even

the small network variations can cause more delay cycles and more software delays in the calculation of FTT.

Hence referring to the figure 6.7, the value of FTT for Silver user profile can be calculated with the help of equation 6.1 as:

FTT (S) = 1818 + 250 * (Ni +1) + 50 * (Ni + 1)
where $FTT(G) = 1818$ ms,
$DL(S) = 250$ ms (according to table 6.1) ,
and $Ds = 50$ ms (The software delay is measured for the experimental setup explained in section 6.3.1)

Finally, calculating Ni from the above formula:
$Ni = 1818 + 250 + 50/3000 = 0$

Thus, $FTT(S) = 2118$ ms

It is important to note that the theoretical value of FTT ( 2118 ms ) is somewhat different from the actual computed value (3565 ms) in figure 6.7. This difference is caused by the network variations on the testbed network, which is denoted by Vnet in equation 6.1.

Equation 6.1 holds true in order to compute FTT in Bronze user case, but the delay value will be changed from 250 ms to 550 ms according to table 6.1. Hence, again referring to figure 6.7, the FTT(B) can be computed in a similar way as equation 6.1:

$$FTT(B) = FTT(G) + DL(B) * (Ni + 1) + Ds * (Ni + 1) \tag{6.2}$$

FTT (B) = 1818 + 550 * (Ni +1) + 50 * (Ni + 1)
where $FTT(G) = 1818$ ms,
$DL(B) = 550$ ms (according to table 6.1) ,
and $Ds = 50$ ms (The software delay is measured for the experimental setup explained in section 6.3.1)

Finally, calculating Ni from the above formula:
$Ni = 1818 + 550 + 50/3000 = 0$

Thus, $FTT(B) = 2418$ ms

Equation 6.1 still holds true for computing FTT in Jitter conditions, but the delay is a random variable in this case with the possible outcome in the range specified in the table 6.1. A more detailed analysis of FTT in Jitter conditions is presented in the following section.

#### 6.3.3.2 FTT Behavior for Different Network Conditions

For a given reference model of Gold user profile, the value of FTT for Fixed packet delay and for Jitter network condition should be ideally same since there is no network delay and variations assumed. However, in the given simulation setup there is a possibility of slight variation in the FTT values due to software delay variations of both fixed delay function and a random delay function. Here, one can argue that there is no need of executing delay functions in the Gold user case. But, on the contrary, the delay functions are still executed to produce an initial delay of '0' value in the given simulation setup. Therefore, the FTT relation for different network settings can be derived as:

$$FTT(J) = FTT(D) + Vsoft \qquad \text{Gold User Profile} \tag{6.3}$$

where, $FTT(J)$ is the FTT value in Jitter network condition,
$FFT(D)$ is the FTT value for fixed packet delay network condition
and $Vsoft$ is the delay variations in executing the two software functions.

The above relationship can be validated from the experimental results shown in figures 6.4 and 6.9. On the contrary, for Silver and Bronze user profiles the FTT(J) and FTT(D) relationship is highly dependent on the sum of the individual delays for a given FTT session produced by the random generator function for Jitter network condition. For instance assuming the Vsoft as '0' for real network case, if the sum of individual random delays is equal to the sum of the individual fixed delays then the value of FFT (J) and FFT (D) will be the same. Similarly, the FFT (J) will be greater or less than FFT (D) if the sum of individual random delays would be greater or less than the sum of the individual fixed delays. Hence, this relationship can be mathematically modeled as:

$$FTT(J) \begin{cases} > FTT(D), if(\sum_{x=1}^{n} DLf_x > \sum_{x=1}^{n} DLj_x) \\ < FTT(D), if(\sum_{x=1}^{n} DLf_x < \sum_{x=1}^{n} DLj_x) \\ = FTT(D), if(\sum_{x=1}^{n} DLf_x = \sum_{x=1}^{n} DLj_x) \end{cases} \quad \text{Silver User Profile} \tag{6.4}$$

$$FTT(J) \begin{cases} > FTT(D), if(\sum_{x=1}^{n} DLf_x > \sum_{x=1}^{n} DLj_x) \\ < FTT(D), if(\sum_{x=1}^{n} DLf_x < \sum_{x=1}^{n} DLj_x) \\ = FTT(D), if(\sum_{x=1}^{n} DLf_x = \sum_{x=1}^{n} DLj_x) \end{cases} \quad \text{Bronze User Profile} \tag{6.5}$$

### 6.3.4 Multimedia Streaming Evaluation

As mentioned in section 5.5, this section presents the evaluation and analysis of video streaming time (VST) of two different videos for different user profiles. Since multimedia streaming services are highly delay sensitive services, the Jitter network condition is further divided into two categories: Periodic Jitter, where a Jitter is produced every time after a fixed period of time throughout the video streaming. Non-Periodic Jitter, represents a Jitter which is produced every time after a variable period of time throughout the video streaming. Table 6.2 presents how video streaming experiments are performed for different user profiles.

#### 6.3.4.1 Video-1 VST Evaluation

Figures 6.14, 6.15 and 6.16 depict the measurement results of VST for Gold, Silver and Bronze user profiles respectively. In case of Gold user profile, the mean VST in Periodic Jitter condition is slightly greater than that of Fixed delay condition. Similarly, the mean VST in Non-Periodic Jitter condition is greater than that of Periodic Jitter condition. As in principle, the Gold user profile gets no ('0') network delay in any network condition, hence, the differences here in streaming times will be assumed as software delays i.e., the computation of random generator function before every delay.

| User Profile | Fixed Delay | Periodic Jitter | Non-Periodic Jitter |
|---|---|---|---|
| Gold User | Delay = 0 | Min Delay = 0<br>Max Delay = 0 | Min Delay = 0<br>Max Delay = 0<br>Min Period = 0<br>Max Period = 0 |
| Silver User | Delay = 150 ms | Min Delay = 100 ms<br>Max Delay = 200 ms | Min Delay = 100 ms<br>Max Delay = 200 ms<br>Min Period = 2000 ms<br>Max Period = 3000 ms |
| Bronze User | Delay = 350 ms | Min Delay = 300 ms<br>Max Delay = 400 ms | Min Delay = 300 ms<br>Max Delay = 400 ms<br>Min Period = 2000 ms<br>Max Period = 3000 ms |

TABLE 6.2: Video streaming experiments

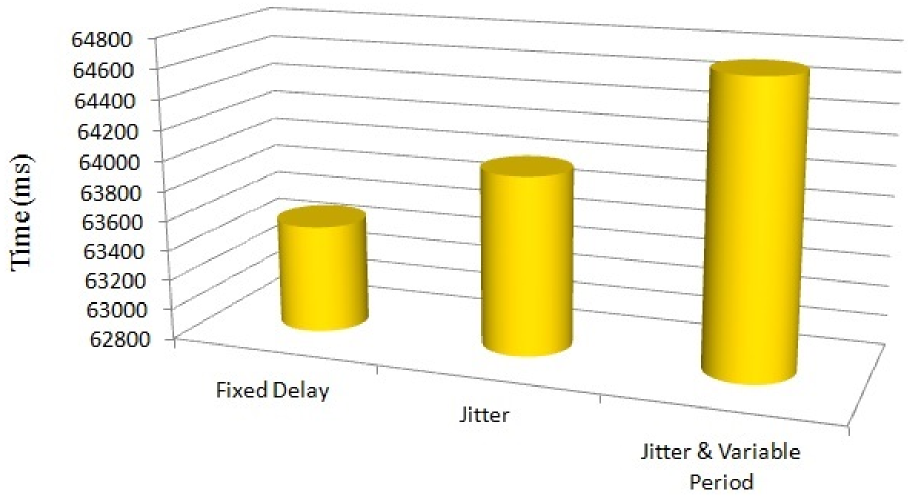



FIGURE 6.14: Video 1 - VST results for Gold users

For Silver and Bronze users, the VST behavior reflects the worst network condition, where Jitter (either Periodic or Non-Periodic) is greater than Fixed network delay. However, it is important to note that there is no significant difference in the computed VSTs in Periodic and Non-Periodic Jitter conditions. This is because of the reason that the period value is randomly generated as closer to 3 seconds (same as Periodic condition) most of the times for Non-Periodic condition.

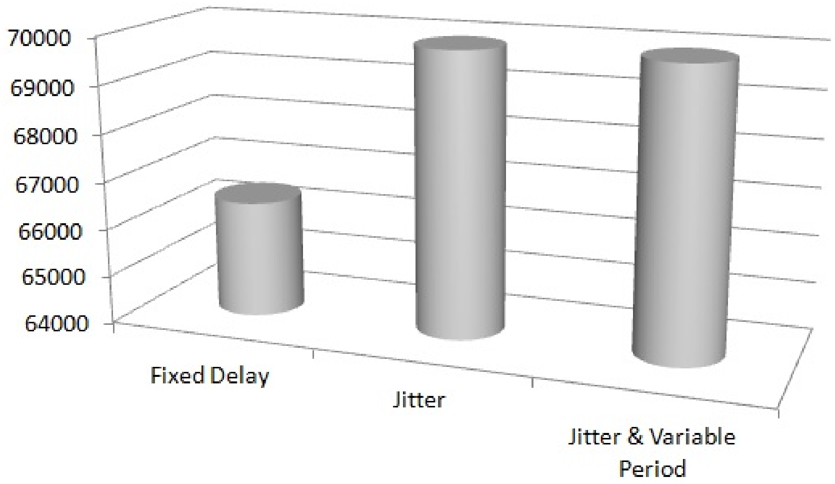



FIGURE 6.15: Video 1 - VST results for Silver users

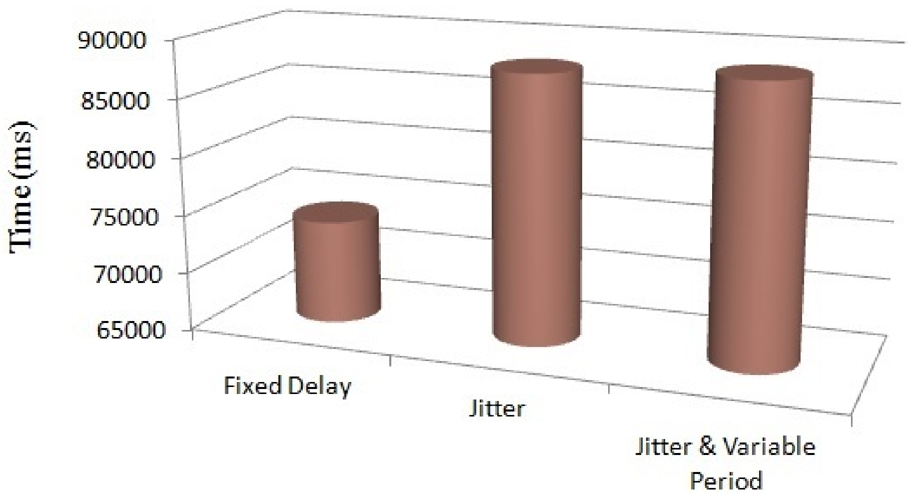



FIGURE 6.16: Video 1 - VST results for Bronze users

#### 6.3.4.2 Video-1 VST Evaluation Based on User Profiles

Figures 6.17, 6.18 and 6.19 present the comparison of mean VST measurements for different user profiles. For a given 45 seconds video in Fixed delay condition, the streaming takes 49,2 seconds for Gold users 52,2 seconds for Silver users and 55,8 seconds for Bronze users. Hence, the average time difference between Gold and Silver users is 5 seconds and between Silver and Bronze users is 6 seconds. For Periodic Jitter case, the streaming takes 49,2 seconds for Gold users 54,6 seconds for Silver users and 63,6 seconds for Bronze users. Hence, the average time difference between Gold and Silver users is 9 seconds and between Silver and Bronze users is 15 seconds. For Non-Periodic Jitter case, the streaming takes 49,2 seconds for Gold users 54 seconds for Silver users and 63,6 seconds for Bronze users. Hence, the average time difference between Gold and Silver users is 8 seconds and between Silver and Bronze users is 14 seconds.

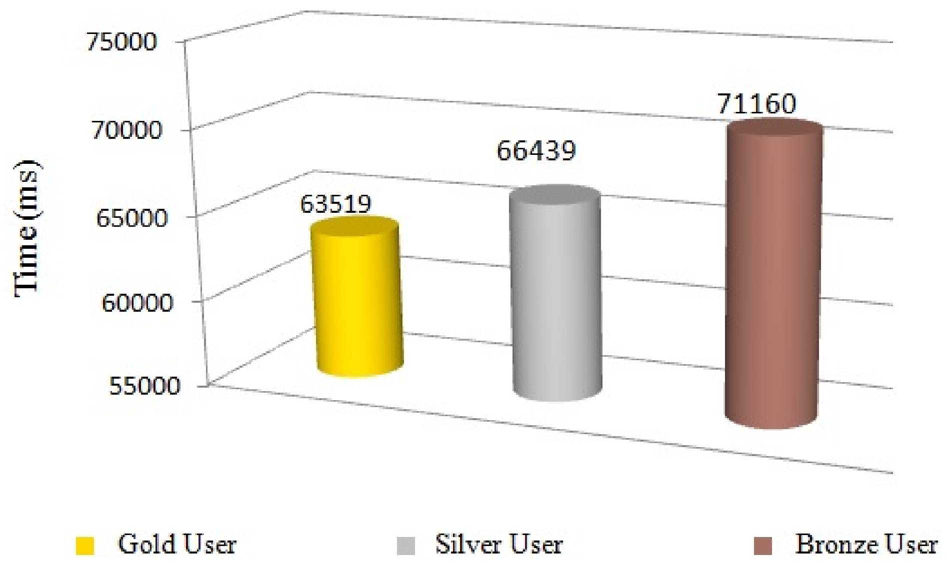



Figure 6.17: Video 1 - VST results in fixed delay network condition

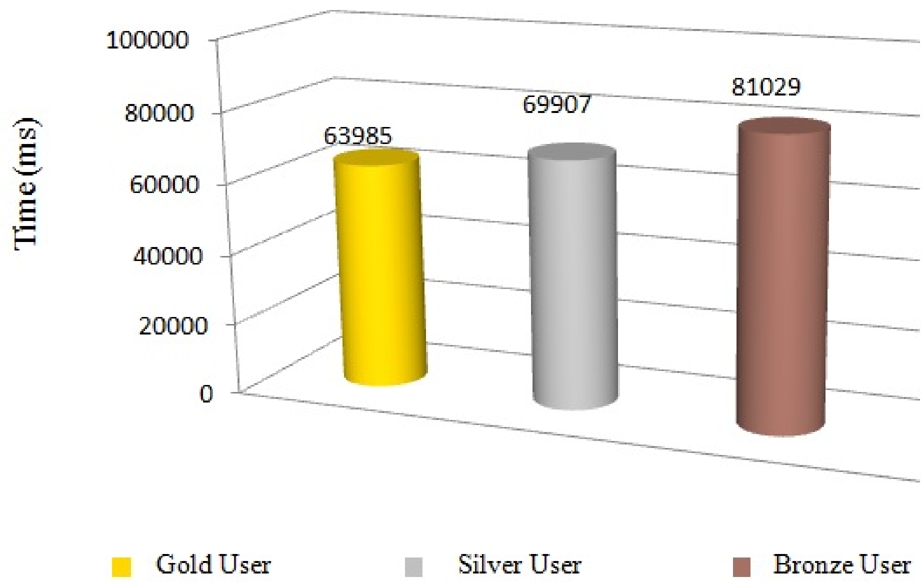



Figure 6.18: Video 1 - VST results in Periodic Jitter network condition

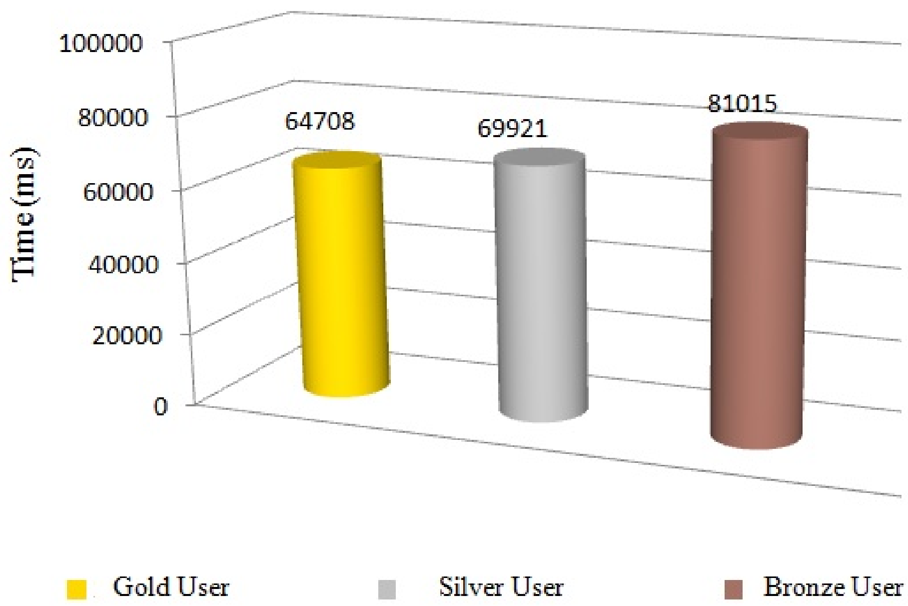



FIGURE 6.19: Video 1 - VST results in Non-Periodic Jitter network condition

It is important to note that for Periodic and Non-Periodic Jitter cases, the video streaming takes same duration of time for corresponding Gold, Silver and Bronze user profiles.

#### 6.3.4.3 Video-2 VST Evaluation

The same experiments were performed to study the VST behavior of Video 2 with same network settings and conditions. Figures 6.20, 6.21 and 6.22 present the results for Gold, Silver and Bronze user profiles in this regard. Similar to the VST results for Video 1 for Gold user case, the mean VST in Periodic Jitter condition is greater than that of Fixed delay condition. For Silver and Bronze users, the VST behavior reflects the worst network condition, where Jitter (either Periodic or Non-Periodic) is greater than Fixed network delay.

#### 6.3.4.4 Video-2 VST Evaluation Based on User Profiles

As in section 6.3.4.4 for Video 1, Figure 6.23, 6.24 and 6.25 present the comparison of mean VST measurements for different user profiles. For a given 1 minute video in Fixed delay condition, the average time difference between Gold and Silver users

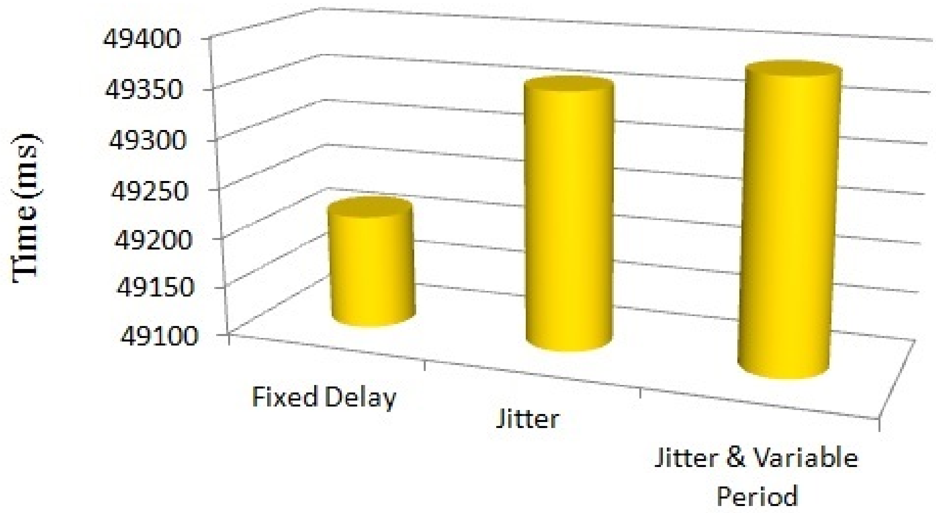



FIGURE 6.20: Video 2 - VST results for Gold users

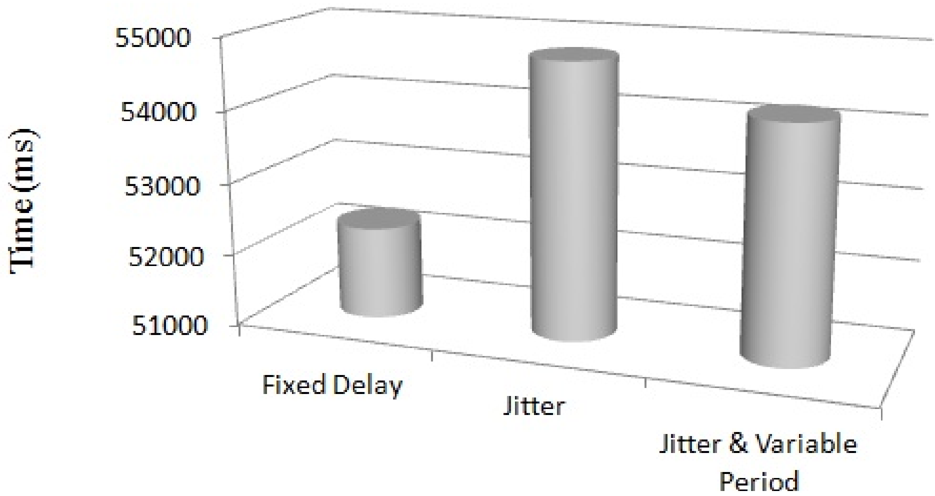



FIGURE 6.21: Video 2 - VST results for Silver users

is 6 seconds and between Silver and Bronze users is 7 seconds. For Periodic Jitter case, the average time difference between Gold and Silver users is 11 seconds and between Silver and Bronze users is 18 seconds. For Non-Periodic Jitter case, the average time difference between Gold and Silver users is 10 seconds and between Silver and Bronze users is 18 seconds.

Here, the VST results obtained for video 2 validates the earlier finding of Video 1 that the video streaming duration is same in Periodic and Non-Periodic Jitter conditions for corresponding Gold, Silver and Bronze user profiles.

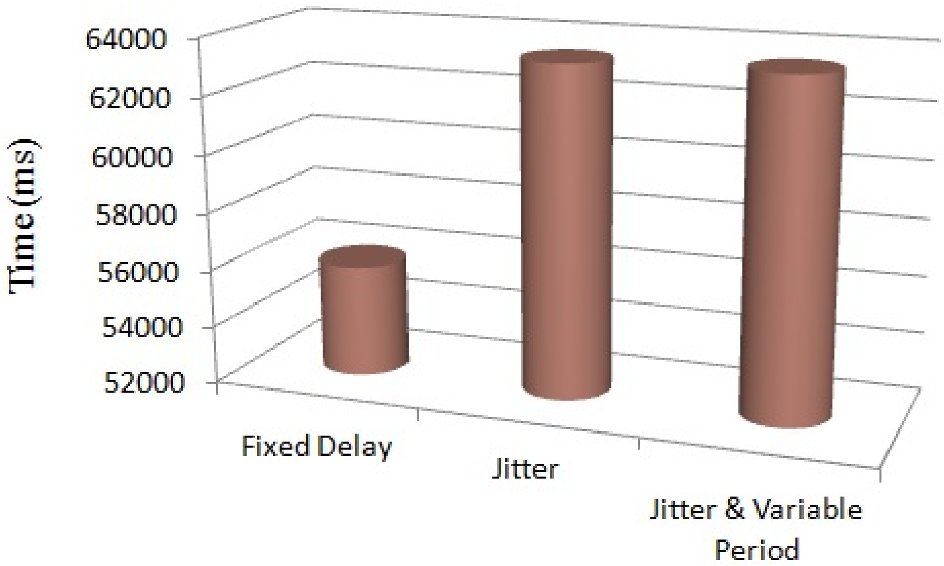



FIGURE 6.22: Video 2 - VST results for Bronze users

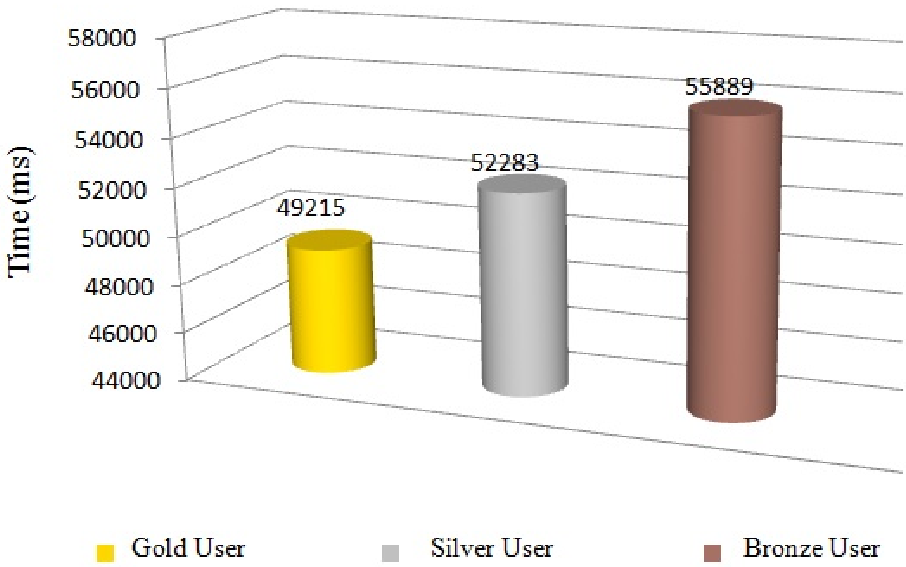



FIGURE 6.23: Video 2 - VST results in fixed delay network condition

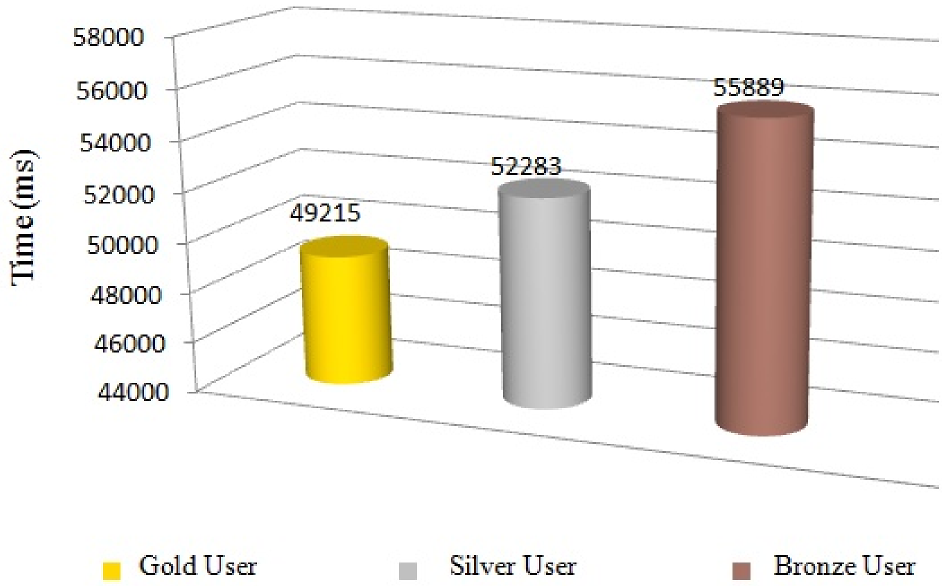



FIGURE 6.24: Video 2 - VST results in Periodic Jitter network condition

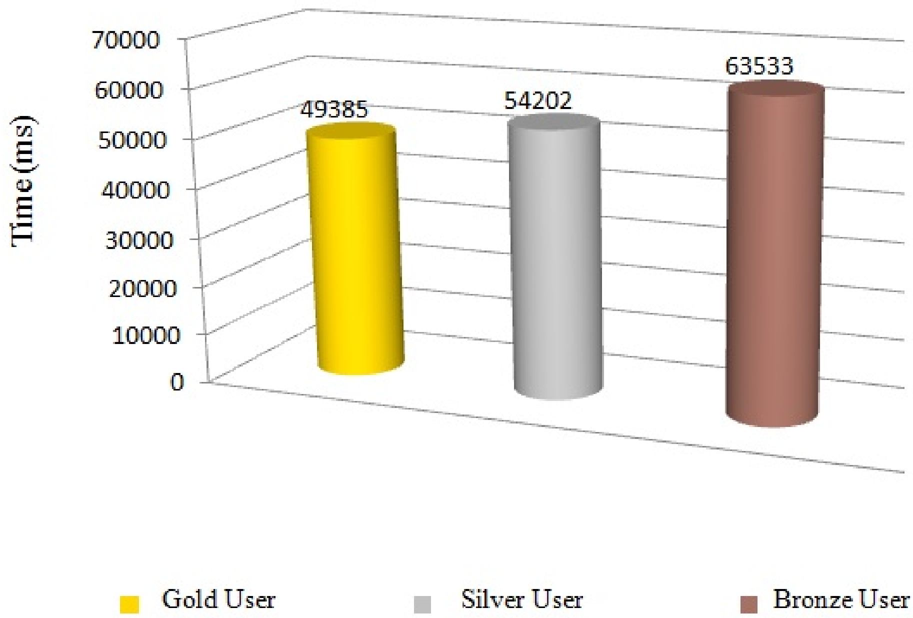



FIGURE 6.25: Video 2 - VST results in Non-Periodic Jitter network condition

### 6.3.5 VST Mathematical Modeling

Similar to file sharing applications, the evaluation of video streaming application is presented in two different modes:

**VST Comparison:** VST evaluations for two different files in order to study the comparison of VST behavior for three different user profiles for a given network condition.
**VST Behavior:** VST evaluations for two different files in order to study the VST behavior of each individual user profile based on the given network condition i.e., Fixed packet delay, Periodic Jitter and Non-Periodic Jitter.

In the subsection below, the mathematical models are presented for two different modes of VST evaluation. The overall objective is to validate the obtained experimental results in the above section and to propose a sophisticated mathematical tool that may help the network operators to study the VST behavior for different network settings and for various user profiles.

#### 6.3.5.1 VST Comparison Based on User Profiles

Let us assume the Gold user profile as a reference model for our VST evaluations i.e., no network delay and network variations for a given VST session. The rest of the user profile models (such as Silver and Bronze profiles) can be easily derived with respect to the reference model. In the simulation setup used as a testbed for underlying network, the same network settings are modeled for Fixed packet delay case as presented in File sharing application in section 6.4.1:

Hence, for a given VST reference model the VST(S) for Silver user profile can be derived as:
$VST(S) = VST(G) + DLi + DL(S) * Ni + DSi + Ds * Ni + Vnet$
where, $VST(G)$ is VST computed for Gold user profile,
$DLi$ is the initial delay induced in the network,
$DL(S)$ is the fixed interval delay,
$DSi$ is the software delay induced by the initial delay generation function,
$Ds$ is the software delay induced by the fixed delay generation function,
$Vnet$ is the delay produced by the network variations or fluctuations for an ongoing

session,
and $Ni$ is the number of iterations how many times a fixed delay is produced for a complete VST span.

In the measurement experiment, the length of the initial delay is kept equal to the fixed interval delays for simplicity i.e., DLi = DL(S) and DSi = Ds. So, simplifying the above equation:
$VST(S) = VST(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) + Vnet$
and

$$VST(S) = VST(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) \qquad (6.6)$$

where, Vnet can be assumed as '0' if a dedicated path (with no fluctuations) is reserved for a particular service session.

The only thing which is unknown in the above equation is the Ni. However, as we know that the delay function is called after every 3 seconds, so the value of Ni can be easily found with the following formula.

$Ni = VST(G) + DL(S) + Ds + Vnet/3000$
where Ni is an integer i.e., the division sign is for integer division (non-fraction). It is worthy to note that Ni is dependent on the network variations Vnet. That means, even the small network variations can cause more delay cycles and more software delays in the calculation of VST.

Hence referring to the figure 6.17, the value of VST for Silver user profile can be calculated with the help of equation 6.6 as:

$VST(S) = VST(G) + 150 * (Ni + 1) + 50 * (Ni + 1)$
where $VST(G) = 63519$ ms, taken from figure 6.17
$DL(S) = 150$ ms (according to table 6.2) ,
and $Ds = 50$ ms (The software delay is measured for the experimental setup explained in section 6.3.1)

Finally, calculating Ni from the above formula:
$Ni = 63519 + 150 + 50/3000 = 21$

Thus, $VST(S) = 67919$ ms
It is important to note that in contrast to the file sharing application, the experimental measurement of video streaming are closer to the corresponding theoretical results.

This is an indication of less variations produced by the underlying network in streaming case.

Equation 6.6 holds true in order to compute VST in Bronze user case, but the delay value will be changed from 150 ms to 350 ms according to table 6.2. Hence, again referring to figure 6.17, the VST(B) can be computed in a similar way as equation 6.6:

$$VST(B) = VST(G) + DL(B) * (Ni + 1) + Ds * (Ni + 1) \tag{6.7}$$

$VST(B) = 63519 + 350 * (Ni + 1) + 50 * (Ni + 1)$
where $VST(G) = 63519$ ms,
$DL(B) = 350$ ms (according to table 6.2) ,
and $Ds = 50$ ms (The software delay is measured for the experimental setup explained in section 6.3.1)

Finally, calculating Ni from the above formula:
$Ni = 63519 + 350 + 50/3000 = 21$

Thus, $VST(B) = 72319$ ms

Equation 6.6 still holds true for computing VST in Periodic Jitter conditions, but the delay is a random variable in this case with the possible outcome in the range specified in the table 6.2. For Non-Periodic Jitter case, equation 6.5 is also true with random delay variable. Moreover, Ni will also become a random variable for Non-Periodic case because the delay interval will be randomly generated according to the table 6.2.

#### 6.3.5.2 VST Behavior for Different Network Conditions

For a given reference model of Gold user profile, the value of VST for Fixed packet delay and for Periodic Jitter network condition should be ideally same since there is no network delay and variations assumed. However, in the given experimental setup there is a possibility of slight variation in the VST values due to software delay variations of both fixed delay function and a random delay function. Here, one can argue that there is no need of executing delay functions in the Gold user case. But, on the contrary, the delay functions are still executed to produce an initial delay of '0' value in the given simulation setup. Therefore, the VST relation for different network settings can be derived as:

$$VST(PJ) = VST(D) + Vsoft \qquad \text{Gold User Profile} \tag{6.8}$$

where, $VST(PJ)$ is the VST value in Periodic Jitter network condition,
$VST(D)$ is the VST value for fixed packet delay network condition
and $Vsoft$ is the delay variations in executing the two software functions.

The above relationship can be validated from the experimental results shown in figures 6.14 and 6.20. On the contrary, for Silver and Bronze user profiles the VST(PJ) and VST(D) relationship is highly dependent on the sum of the individual delays for a given VST session produced by the random generator function for Periodic Jitter network condition. For instance assuming the Vsoft as '0' for real network case, if the sum of individual random delays is equal to the sum of the individual fixed delays then the value of VST (PJ) and VST (D) will be the same. Similarly, the VST (PJ) will be greater or less than VST (D) if the sum of individual random delays would be greater or less than the sum of the individual fixed delays. Hence, this relationship can be mathematically modeled as:

$$VST(PJ) \begin{cases} > VST(D), if(\sum_{x=1}^{n} DLf_x > \sum_{x=1}^{n} DLpj_x) \\ < VST(D), if(\sum_{x=1}^{n} DLf_x < \sum_{x=1}^{n} DLpj_x) \\ = VST(D), if(\sum_{x=1}^{n} DLf_x = \sum_{x=1}^{n} DLpj_x) \end{cases} \qquad \text{Silver User Profile} \tag{6.9}$$

$$VST(PJ) \begin{cases} > VST(D), if(\sum_{x=1}^{n} DLf_x > \sum_{x=1}^{n} DLpj_x) \\ < VST(D), if(\sum_{x=1}^{n} DLf_x < \sum_{x=1}^{n} DLpj_x) \\ = VST(D), if(\sum_{x=1}^{n} DLf_x = \sum_{x=1}^{n} DLpj_x) \end{cases} \qquad \text{Bronze User Profile} \tag{6.10}$$

## 6.4 Conclusion

The chapter presents the design and implementation of M2M QoS framework and explains how it helps to control the QoS level for an ongoing end-to-end service session. The former research of MSP [9] presents the QoS management of mobile web services as a part of agreement evaluation process of its proposed SLA framework. For each new service deployed in the system, a QoS handler is written by the service provider or sub-contracted third parties, which is kept responsible to monitor QoS parameters negotiated in the SLA agreement. In general, the proposed agreement evaluation process conforms to the standard of the Web Services Agreement Specification (WS-Agreement) [18] for mobile terminals. Whereas, this dissertation presents a QoS control and management framework for M2M services that conforms to the standards of 3GPP Policy and Charging Control (PCC) architecture used in LTE-EPC [38]. Some salient features of the proposed QoS framework are as follows:

- The HTTP digest authentication is implemented for the authorized registration of users in the IMS network.
- The proposed QoS framework exercises control at every stage of service provisioning mechanism i.e., user registration, session establishment and media delivery phase.
- The proposed framework enables the cellular operators to employ QoS control based on the user profiles classification. Three different user profiles (Gold, Silver and Bronze) are created to study the behavior of file sharing and video streaming applications for different network settings.The mathematical models for FTT (File Transfer Time) and VST (Video Streaming Time) are developed for each user profile under different network conditions.

It is concluded that a for a given reference model of FTT i.e., FTT(G) for instance, the corresponding FTT models such as, FTT(S) and FTT(B) can be easily derived based on the following equation:

$FTT(S) = FTT(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) + Vnet$

$FTT(B) = FTT(G) + DL(B) * (Ni + 1) + Ds * (Ni + 1) + Vnet$

Similary, VST(S) and VST(B) can be derived from the following formulation:
$VST(S) = VST(G) + DL(S) * (Ni + 1) + Ds * (Ni + 1) + Vnet$
$VST(B) = VST(G) + DL(B) * (Ni + 1) + Ds * (Ni + 1) + Vnet$

It is identified that there is a slight difference between the theoretical values computed from the mathematical models and the corresponding experimental computed values. However, this difference is caused by the network variations on the testbed network, which is denoted by Vnet in the above equations.

# Chapter 7

# Conclusions

The research work is aimed to contribute in the cellular world with a new paradigm of so-called Mobile-to-Mobile service networks. The basic idea is how mobile devices can collaborate and disseminate information in a service-oriented fashion over the advanced cellular data networks. In past, such kind of service networks were only limited to local area networks, such as WiFi and bluetooth etc. However, realizing the same concept for cellular networks is not straight forward and pose some of the following challenges:

**Quality of service and policy management** In a typical Internet paradigm, the networks are aimed to offer a best effort quality of service to their customers. In order words, the customers will get a good quality network in the events of not much traffic over the network. But, there are no quality of service guarantees in otherwise situations. Thus, considering the bandwidth hungry nature of multimedia services and bandwidth restrictions of cellular networks, the ordinary best effort policy seems not to comply with the requirements. Therefore, a special policy control mechanism is necessary to introduce for the provisioning of M2M service networks for cellular networks.

**Service Blocking** In order to avoid OTT traffic over their networks, the cellular operators usually block the IP access of their consumer mobile devices through the data network. Thus, an M2M interaction is not possible without investigating and integrating a standardized communication subsystem in this regard.

**Standardized Mobile Server platform** The Mobile Server Platform introduced in [9] does not conform to the standards of cellular industry and lacks the following required features:

- It does not specify a standardized way to register itself as a client or an application server with the cellular data network.
- It does not define the service discovery and service invocation mechanisms by using the standardized interfaces and protocols of the cellular system.
- It does not integrate with the standardized policy and charging solution of an operator data network.

In order to address these challenges, the dissertation contributes a Mobile Server Platform that has the ability to register itself in the standardized application subsystem of 4G / LTE cellular network, named as IP Multimedia Subsystem. The introduction of IMS subsystem has enabled the cellular operators to host applications in the LTE network, control them and apply their QoS and charging policies accordingly.

Thus, based on the standardized communication interfaces introduced by 3GPP for IMS and LTE-Evolved Packet Core mobile system, the M2M services like multimedia instant messaging and real-time audio video streaming are developed for Mobile Server Platform, which are fully compatible to work with Ericsson IMS and Fraunhofer OpenIMS, OpenEPC testbeds. In order to address the QoS concerns, QoS framework has been introduced in order to fulfill the requirements of 3GPP Policy and Charging Control (PCC) mechanism. The last but not the least, the performance evaluation of Mobile Server Platform is extensively carried out based on the various user classifications and service types.

## 7.1 Limitations and Outlook

Although the research has reached its aims, this section is dedicated to discuss the overall implications and limitations of the proposed work. Moreover, a potential road-map for the future work is also presented in this discussion.

First, from the implementation point of view, all the implementations and testing carried out in this work are compliant and limited to the IMS testbeds available from Ericsson and Fraunhofer institutes running in a lab environment with limited resources and infrastructure. Therefore, it is expected that if tested in the real world scenario, the current implementation of Mobile Server Platform may require some system level modifications prior to testing with the operator's LTE and IMS infrastructure. For instance, the proposed server platform only implements a commonly used authorization scheme for SIP communication i.e., HTTP Digest Access Authentication. Thus for security reasons, if an operator's network does not support this authentication scheme then the server platform will not be able to gain access to the operator's IMS network.

Next, in the proposed architecture the registration of users' devices as mobile application server requires a manual assignment of static IP addresses (may not be available from network operators owing to business nodes) and Port numbers against every individual device in the operator's DNS system. This could be a tiresome process for a network of several hundred devices or more. Thus, a better and more flexible approach would be the automatic assignment of static IPs to particular user profiles. That means, whenever a certain user is logged into the operator's data network, the network assigns the same IP address every time.

Furthermore, chapter 6 presents a schematic study of the file transfer and video streaming latencies for various types and characteristics of the underlying network i.e., fixed packet delay, periodic and non-periodic Jitter conditions etc. Whereas, the evaluation results obtained in this regard do not reflect the real-time testings on operator network rather are based on the software simulations of such network types and conditions. Similarly, the corresponding mathematical models are developed in view of such software implementations of different network behaviors.

In addition to addressing the above mentioned limitations of our work, following improvements and extensions to the M2M service model are also recommended from the futuristic research and development point of view:

The data center network is the backbone of every telecom network. Whereas, the relentless growth of multimedia traffic has seriously questioned the rigorous architecture of legacy data centers. Hence, Software-Defined Networking (SDN) is a new networking paradigm that decouples the control plane of a data center network from its user plane and offers programmability of data plane devices to manage and control the ongoing traffic flows and ensure application-aware quality of service on top. Therefore, in order to enhance the dynamicity and control of the proposed telco-oriented M2M service model, an integration with state-of-the-art SDN networking paradigm can help in the optimization of the bursty M2M traffic. Whereas, the load balancing feature of an SDN enabled network further secures the ability to satisfy the different QoS requirements of M2M services by instant traffic identification and dynamic traffic rerouting mechanisms.

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# List of Figures

# List of Tables

# Abbreviations

| | |
|---|---|
| ARP | Address Resolution Protocol |
| AS | Application Server |
| DNS | Domain Name System |
| EPC | Evolved Packet Core |
| HSS | Home Subscriber Server |
| HTTP | Hypertext Transfer Protocol |
| i5 IMS MSP | Informatik 5 IMS Mobile Service Platform |
| I-CSCF | Interrogative-Call Session Control Function |
| iFC | Initial Filter Criteria |
| IMS | IP Multimedia Subsystem |
| KPI | Key Performance Indicator |
| MAS | Mobile Application Server |
| MSRP | Message Session Relay Protocol |
| MPLS | Multiprotocol Label Switching |
| MWS | Mobile Web Services |
| PCRF | Policy and Charging Rules Function |
| PCEF | Policy and Charging Enforcement Function |
| P-CSCF | Proxy-Call Session Control Function |
| QoE | Quality of Experience |
| QoS | Quality of Service |
| RTSP | Real Time Streaming Protocol |
| REST | Representational State Transfer |
| RTCP | Real-Time Transport Control Protocol |
| RTP | Real-time Transport Protocol |

| | |
|---|---|
| S-CSCF | Serving-Call Session Control Function |
| SDP | Session Description Protocol |
| SDS | Service Development Studio |
| SIP | Session Initiation Protocol |
| UA | User Agent |
| UAC | User Agent Client |
| UI | User Interface |
| URI | Uniform Resource Identifier |
| URL | Uniform Resource Locator |
| WLAN | Wireless Local Area Network |

# Appendix

## 7.2 Mobile Server Platform

The below listed steps define the login to the MSP prototype and describes precisely how the user can interact at the first glance with the prototype:

1. The prototype is developed for android Platform with a minimum supporting version of 2.3.3 to the newest android version 4.1.2
2. The MSP framework starts as an application on the phone and it brings an interface (Login Screen) to the user where he/she is asked to provide the login credentials
3. As soon as the client enters the user and password, the credentials are sent to the HSS module of IMS which verifies the existence of the user in the IMS database.
4. The client has to press the enter button
5. SIP register message is send to the IMS framework (in our case first to the P-CSCF)
6. The successful registration will bring a contact list screen to the user as shown in figure 7.1.
7. This is the main interface window of the MSP framework and this enables the client to use different services based on the need.

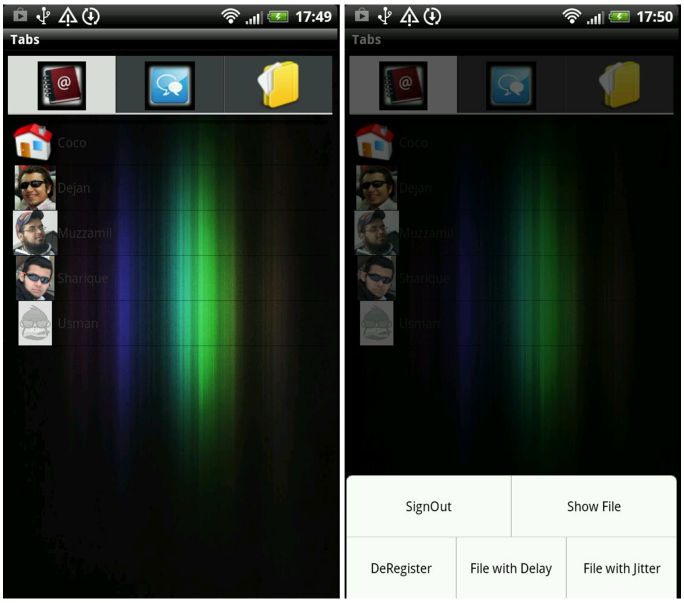


FIGURE 7.1: Mobile Service Platform - Contacts list Interface

### 7.2.1 Multimedia Streaming Service

How the streaming service is working in MSP Framework is described in below mentioned steps:

1. First, the user selects a contact from the main interface and then there appears one interface dialog box containing the list of services a user can select. The figure 7.2 shows the screen view of the contact options dialog.

2. The user then selects the option for Audio/Video Streaming Service which enables us to run this application.

3. On the client side the application will bring the an interface which you can see on the right-side screenshot in figure 6.2

4. After the user click on start a SIP request initiated and after being authenticated and successful end-to-end connection establishment, the user will be able to see the live video on his android device. The screen reflects that the user can also have the possibility to Pause, Stop and Replay the video.

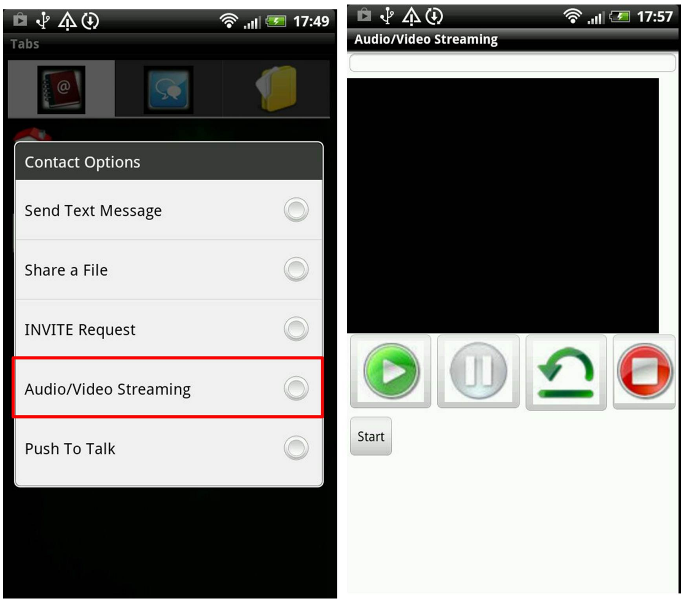


FIGURE 7.2: Mobile Server Platform - Audio/Video Streaming Service Interface

### 7.2.2 File Sharing Service

1. The user will select the service of File Sharing from the contact options dialog "Share a File", the above figure shows the dialog box

2. Then the interface will bring the user to the select a file from the Gallery. The user can select the picture from any album which he would like to share as shown in figure 7.3.

3. Afterwards, the receiver will acknowledge one time that he really wants to have the file.

4. After a successful transfer of file, the user can see right away the received file. Moreover, the application stores the file in the memory so that the user can also view in later.

### 7.2.3 Text Messaging Service

The user who wants to send the other user will use the application option for "Send Text Message" and then type in the message which he wants to send to the other user. The figure 7.4 show the GUI of the IM service.

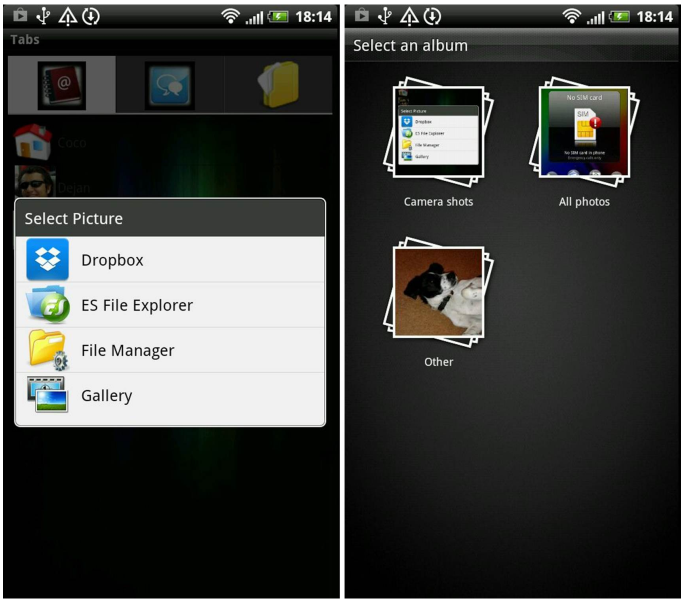


FIGURE 7.3: Mobile Server Platform - File Sharing Service Interface

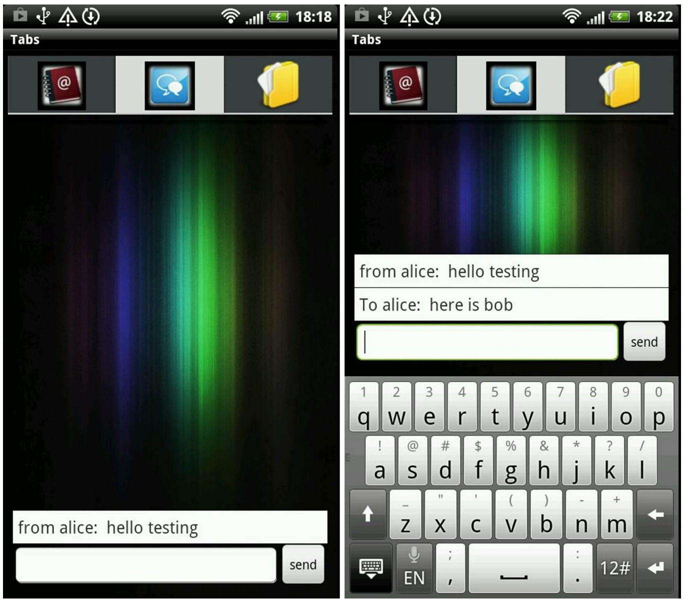


FIGURE 7.4: Mobile Server Platform - Text Messaging Service Interface

## 7.3 FOKUS IMS Environment Setup

Linux version setting 12.10

The current free open source version of FOKUS OpenIMS Core is installed on Ubuntu 12.04 operating system.

The table A.1 shown below reflects the important terminal nodes of the IMS testbed against their corresponding port numbers for communication purposes.

| Element | Port Number |
|---|---|
| P-CSCF | 4060 |
| I-CSCF | 5060 |
| S-CSCF | 6060 |
| Diameter | 3868, 3869, 3870 |

Table 7.1: Terminal nodes port numbers [50]

## 7.4 HSS Configuration [Specific Settings done for FOKUS OpenIMS Core]

The Web Interface of OpenIMS Core lists all the operational and management functionalities including the creation of new User, Application Server, Filter Criteria and Trigger point. The table A.2 illustrates the web interface address along with the credentials.

| |
|---|
| To access the OpenIMS Core Web Interface, use "**http://localhost:8080**"<br>Username: hssAdmin<br>Password: hss |

Table 7.2: OpenIMS Core - HSS Web Interface [50]

## 7.5 The PCRF Terminal Setup

For the Policy management in our thesis work we have also performed some configuration to run the PCRF terminal. The table A.3 below shows some of the very important changes of UCT IMS Policy framework required by the testbed in order to support QoS for a particular user. All these changes are done in 'pcscf.qos.cfg' file

## 7.6 PCRF Web Interface Setup

The Web Interface is used in order to review and manage the user policies and codecs information which enable the user permission to the demanded QoS. With the intention of accessing the PCRF web interface first the PCRF and PCEF terminal

```
loadmodule "/opt/OpenIMSCore/ser_ims/modules/cdp/cdp.so"
 //to be customized with your pcscf.xml path
modparam("cdp", "config_file", "/opt/OpenIMSCore/ser_ims/cfg/pcscf.xml")
loadmodule "/opt/OpenIMSCore/ser_ims/modules/cdp_avp/cdp_avp.so"
modparam("pcscf","use_pcc",1)

modparam("pcscf","forced_qos_peer","pcrf.open-ims.test")
modparam("pcscf","qos_release7",1)
 //to be customized with your IP  address to pcscf
modparam("pcscf","ipv4_for_signaling","192.168.2.101")
modparam("pcscf","port_for_signaling",4060)
```

Table 7.3: OpenIMS Core - Policy Framework settings

needs to be up and running. Below mentioned URL is used to access the Web interface.

| |
|---|
| http://localhost:8180/uct_web_pcm/ |

Table 7.4: OpenIMS Core - HSS Web Interface

The Policies tab states both the Domain Policies as well as the Generic Policies. The current version of UCT PCRF only supports the Domain Policies which contains the detailed list of the Codec IDs, Codec Types and the Media Types of the available and supported codecs. Furthermore, this tab also illustrates the QoS Class and domain Authorization Rules which comprises the Uplink and Downlink bandwidth. The figure A.1 shows the codec details in the domain policies.

The Topology and Resources tab allow us to view the status of the PCRF and PCEF modules. The figure A.2 shows that we have only one PCRF with the Domain 'open-ims.test' and its status is currently ON. Similarly, the figure A.3 projects that PCEF module is also up and running and the Status parameter represents the value to be in ON state.

For more detail about the UCT IMS Client, please find the below link
https://lists.berlios.de/pipermail/uctimsclient-users/2009-May/000562.htm

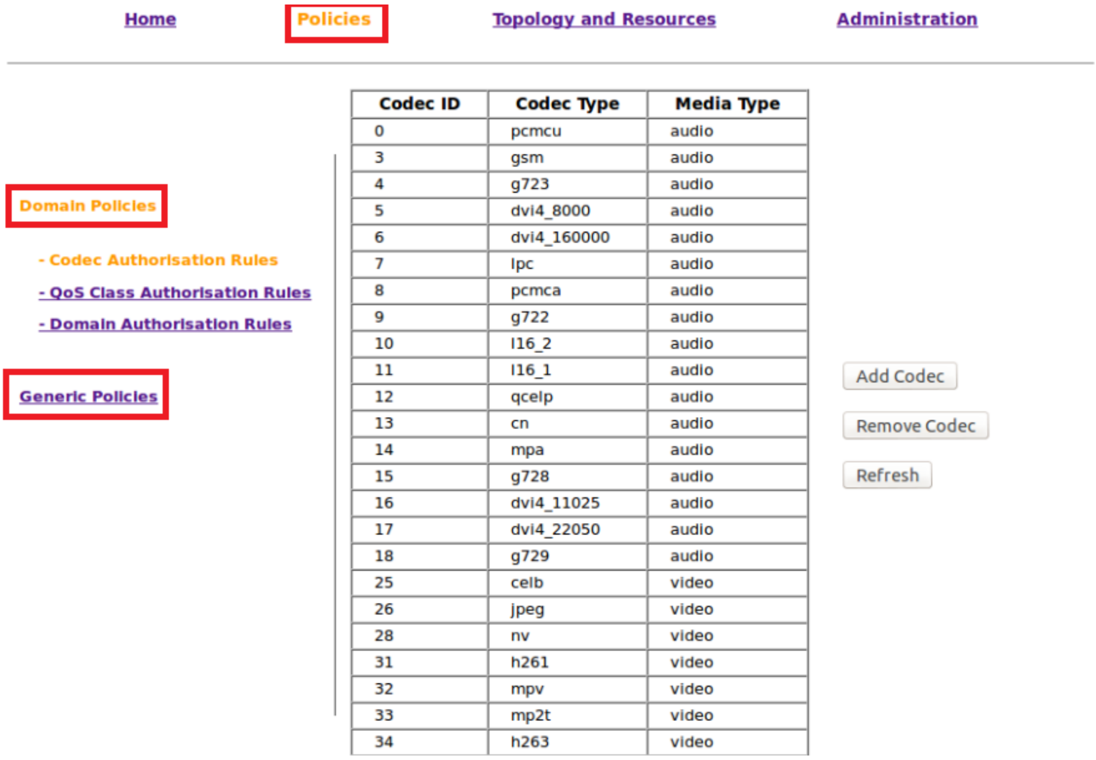


FIGURE 7.5: UCT Policy Control Management System - Domain Policies

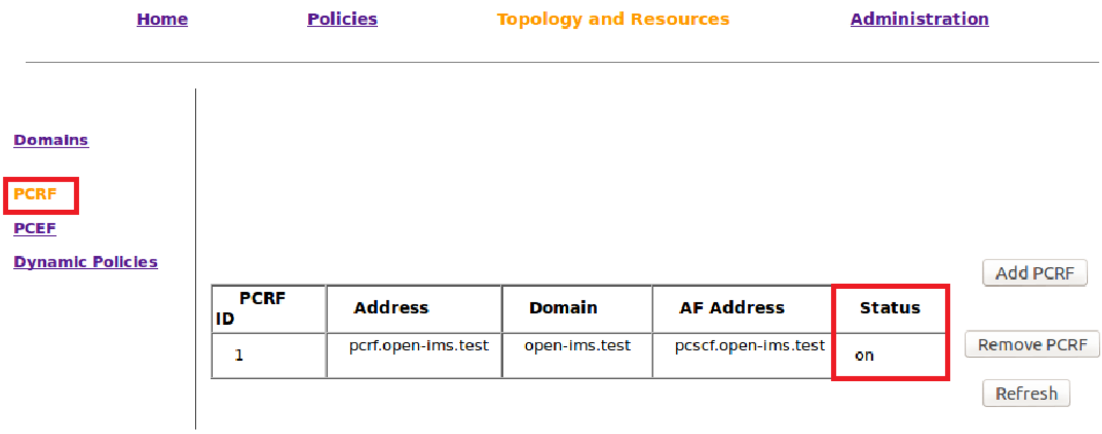


FIGURE 7.6: PCRF status in Web Interface

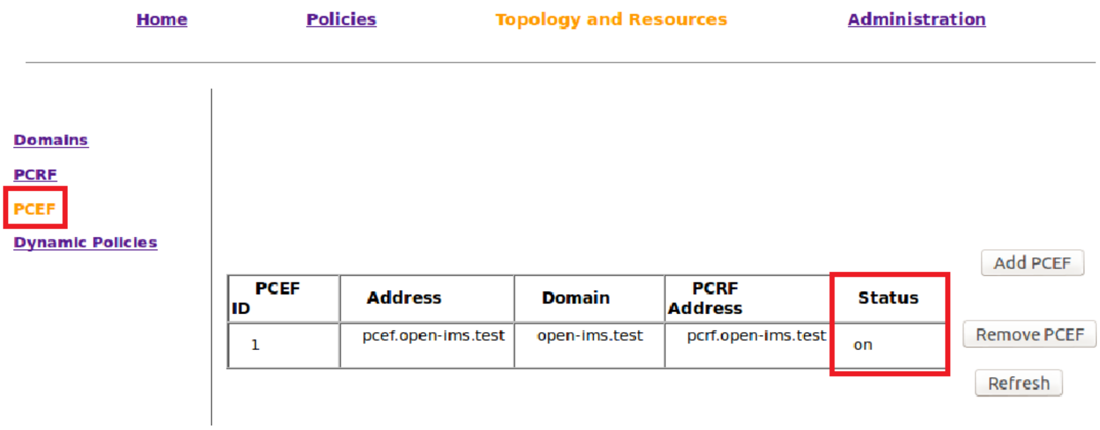


FIGURE 7.7: PCEF status in Web Interface