TECHNISCHE UNIVERSITÄT WIEN

DISSERTATION

Quality-of-Service for Location-Based Services in UMTS

ausgeführt zum Zwecke der Erlangung des akademischen Grades eines Doktors der technischen Wissenschaften unter der Leitung von

o. Univ. Prof. Dr.-Ing. Harmen R. van As
Institut für Breitbandkommunikation

und Zweitbegutachter

Em. o. Univ. Prof. Dr.-Ing. Ernst Bonek
Institut für Nachrichtentechnik und Hochfrequenztechnik

eingereicht an der Technischen Universität Wien
Fakultät für Elektrotechnik und Informationstechnik

von

Dipl.-Ing. Igor Brusic
Matr.-Nr. 9625062
Mengergasse 21/3, 1210 Wien

Wien, im Mai 2005

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Zusammenfassung

Die dritte Generation der Mobilfunknetze, UMTS (Universal Mobile Telecommunication System) soll die Funktionalität von mehreren bestehenden Mobilfunksystemen integrieren, für weltweite Erreichbarkeit über nur ein Gerät sorgen und eine Vielzahl neuer Dienste, die über die Sprachtelephonie hinaus gehen, ermöglichen. Die Basis dafür sind derzeitige Datenraten von bis zu 2 Mbit/s und paketvermittelte Datenübertragung basierend auf dem Internet Protokoll (IP).

Die Verwendung des Internet Protokolls bringt den Vorteil Internet Anwendungen nutzen und sich einfach mit lokalen Netzen auf TCP/IP Basis verbinden zu können, aber auch den grossen Nachteil unterschiedliche Dienste im Netz nicht differenzieren zu können. Das IP Protokoll arbeitet noch immer nach dem Best Effort Prinzip, wodurch Echtzeit- und Nichtechtzeitanwendungen in der Übertragung gleich behandelt werden.

Bezüglich der neuen Dienste werden derzeit hochqualitative Multimedia Dienste in den Vordergrund gestellt. In dieser Arbeit wird aber der Gruppe von Diensten, die als Basis die Position des mobilen Endgerätes nutzen, das viel grössere Potential zugestanden. Anwendungen wie „Friend Finder", „Wo ist das nächste ...?" oder Navigationsanwendungen sind nur ein kleiner Teil der vielen Anwendungen, die folgen könnten.

Eine wesentliche Voraussetzung für den Erfolg von lokalisierungsbasierenden Diensten (Location-Based Services, LBS) und UMTS überhaupt ist die Möglichkeit Dienste in der Übertragung zu priorisieren. Nur dadurch ist es möglich, die Sprachkommunikation und andere Echtzeitanwendungen gegenüber der restlichen Datenkommunikation zu bevorzugen. Die dafür notwendigen Methoden, Mechanismen und Architekturen werden unter dem Begriff Dienstgüte (Quality-of-Service, QoS) behandelt.

Für den LBS Teil werden Positionierungsmethoden und die Standardisierung in GSM, GPRS und UMTS beschrieben. Genauso werden bestehende QoS Mechanismen detailliert dargestellt. Ein grosser Teil dieser Arbeit bezieht

sich auf Scheduling Methoden die wesentlich sind in der gemeinsamen Nutzung von Netzressourcen, besonders bei hohen Auslastungen. Einen besonderen Stellenwert bekommt auch die QoS Methode die derzeit im Internet fast aus-schliesslich verwendet wird. Dies ist die Überbuchung von Übertragungskapazitäten, auch Overprovisioning genannt.

Es werden bestehende QoS Mechanismen der IETF und die 3GPP QoS-Architektur dargestellt. Die erarbeiteten Einsichten führen zu einer neuen QoS Architektur für UMTS, die es ermöglicht, Echtzeitanwendungen wie Sprache zu priorisieren und gleichzeitig LBS Anwendungen differenzieren zu können.

Durch Simulationen wird eindeutig gezeigt, dass Overprovisioning nur begrenzt einsetzbar ist. Solange nur eine schwache Differenzierung zwischen den Anwendungen vorhanden ist, kann auch die Steigerung der Bandbreite im Netz eine Lösung des QoS Problems darstellen. Sobald es aber eine kleine Klasse von Paketen hoher Priorität gibt, sieht die Situation ganz anders aus. In diesem Fall müsste man sehr viel mehr Bandbreite einsetzen, um die selben Verzögerungszeiten wie durch den Einsatz von QoS Mechanismen zu erreichen.

Die simulativen Untersuchungen und die vorgeschlagene QoS Architektur für UMTS sollen somit einen Beitrag für den Erfolg von LBS Anwendungen in UMTS darstellen.

Abstract

The expectation to the third generation of mobile networks, UMTS (Universal Mobile Telecommunication System) is to integrate the functionality of existing mobile systems, provide worldwide coverage with only one device and allow numerous new services beyond the traditional voice communication. These aims are based on today's data rates of up to 2 Mbit/s and on packet switched data transfer based on the Internet Protocol (IP). The use of the Internet protocol has the advantage that Internet applications can be used and the connection with other TCP/IP based networks becomes simple. On the other hand, there is also the disadvantage that the Internet protocol does not differentiate between applications. In the transport of data, the IP protocol still works according to the best effort principle and by this real time and non-real time applications are handled in the same way.

Regarding new services for UMTS, at present high quality multimedia services are pushed in to the foreground. In this work, we concede higher potential to services make use of the mobile equipment's position. Applications like "Friend Finder", "Where is the nearest ?", or navigation solutions are only a small part of what could follow. One of the most important pre-conditions for the success of Location-Based Services (LBS) and UMTS in general, is the possibility to prioritize services during transmission. Only in this way voice communication and other real-time applications can be insulated from the influence of heavy data transmission. The Quality-of-Service (QoS) concept deals with the relevant methods, mechanisms, and architectures.

For the LBS part, we describe all localization methods in discussion today and also cover the standardization efforts for GSM, GPRS, and UMTS. QoS mechanisms are presented in detail. One big part of this work concerns scheduling mechanisms, which are a very important component in the shared use of network resources, especially in times of high utilization. Overbooking of transmission capacities, also called overprovisioning, is handled in detail. This is the QoS method currently used almost exclusively in the Internet.

This work presents existing QoS mechanisms proposed by the IETF and the 3GPP QoS architecture. The insights elaborated, lead to the proposal of a new

QoS architecture for UMTS, which permits to prioritize real-time applications such as voice and at the same time differentiate LBS applications.

Simulation examples presented in this work show that overprovisioning can only be employed within certain boundaries. If there is only little differentiation between applications, an increase of bandwidth in the network can also be a solution to the QoS problem. However, if only a small class of high priority packets exists, the situation changes. Now, a lot more bandwidth would have to be used in order to reach the delay times achieved by the employment of QoS mechanisms.

Performance evaluations of QoS mechanisms and the proposed QoS architecture for UMTS should contribute to the success of LBS applications in UMTS.

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1 Introduction

The topic of this thesis is quality-of-service (QoS) for location-based applications. The importance of this topic lies in the realistic assumption that location based applications will be the main traffic generator in mobile communication networks in the near future but real-time traffic like voice will still be one of the main revenue generators. The underlying transport technology will be based on packet switching and the requested protocol will be the internet protocol. Together this will challenge every mobile network provider. The problem is that packet-switched networks based on the internet protocol do not differentiate real-time traffic from other data traffic and therefore they do not provide guaranteed quality-of-service.

The internet protocol only delivers service based on best effort basis. The common argument is that without being capable of giving QoS guarantees, network providers will not make it to pass the threshold of the number of LBS users, which is necessary for LBS to be successful. The first LBS applications like *Friend Finder*, *Where is the nearest ...?* and *Mapping Applications* can be treated as the top of an iceberg. Today there are umbrella terms like roadside-assistance, traffic and navigation and location-based billing but the existing applications just show a small part of what LBS is capable of and make the user aware of further mobile device capabilities beside classical voice transmission. With a growing number of applications, the relatively small number of users of non-voice applications will soon raise. A growing number of services that can be offered, makes it more probable to find larger numbers of users to whom these services appear attractive[1]. One of the prerequisites is standardized hard- and software, but also increased positioning accuracy. Consequently, more users will employ LBS services, which will generate a large amount of

[1]A good example for this is given by the Japanese mobile operator NTT DoCoMo service called i-mode.

traffic, influencing the quality of real-time applications such as voice. This will again give a leading role to QoS topics. Overloading is seen as one of the possible solutions but also as particularly critical in packet switched networks. The next option to enhance QoS is buffer and queue management together with fairness mechanisms. In general, it can be said that one can only give guarantees if it is possible to keep them. Therefore, also control mechanisms and policing play an important role.

The next problem is that quality-of-service has to be considered end-to-end. For a voice transmission that means from the calling user to the called user. For data transmission, it is between the client and the server. A network consisting of different parts lies between them. The first part is the access part, where the user or the clients access the network, and the second is the transport part, which connects the access part with the third part, the backbone. In any of these parts, quality-of-service has to be supported, but the clue is - with different mechanisms. One part of this work is to define proper QoS mechanisms, resulting with a proposal for a QoS architecture for UMTS.

Another problem is due to the fact that the IP protocol initially was developed for employment in computer networks rather than telecommunication networks. For computer networks it was assumed that all intelligence would be situated within the end devices and the network itself would be more or less "stupid". The opposite is true for telecommunication networks. If we now want telecommunication networks to offer new services based on the IP protocol, the following questions arise: Who will or can develop new services? Who is responsible for controlling a new service? Will this be done by the end user or by the network? Answers to these questions are directly connected to the question whether network providers will offer open interfaces. This means for an "external" LBS application the prime question will be whether it will be given positioning information. This of course is also a question of privacy, but even more of control over certain information. Network providers fear that they might lose control over the services offered and that their networks will only be used for data transmission. This evolvement is the basis for the Internet's success. First steps have been taken in this direction with the application of platforms like CORBA (Common Object Request Broker architecture), which support communication between distributed objects and achieve interworking of distributed systems and processes. Hence, the Internet Inter-ORB Protocol (IIOP) has been specified for access to network resources on top of the TCP/IP protocol.

In the framework of this thesis, different mechanisms for supporting service control and quality-of-service are investigated. In the first place, it concerns the work of standardization organizations. This results in a QoS architecture, which should enable provision of LBS applications to a large number of users. In addition, existing scheduling mechanism are compared and further investigations in *Overprovisioning* are introduced as feasible methods to overcome the QoS problem.

In the first part of this introduction, we will give a more particular consideration on the motivation for location-based services and some non-technical issues concerning the success of new services. Bound to this topic, possible approaches for reaching the threshold of the number of users will be presented. In the Section *Technological Evaluation*, we are pointing out some important technological issues concerning LBS like data rates, positioning methods, terminals, network architecture, and quality-of-service. Afterwards, mechanisms closely tight to the QoS issue with the Internet Protocol are discussed. At the end of the introduction an overview of this thesis is given.

1.1 The Motivation for Location-Based Services

Over the past 10 years international economy has seen an almost unbelievable increase in two sectors: Internet and mobile communication. In 1991, the WTO (World Trade Organisation) noted 4.5 million Internet users, today in the beginning of 2005 there are over a billion users[2]. This increase showed even more drastic with mobile communication. OECD Communication Outlook stated some 11.181 million mobile users in 1990, while there are 1.456 billion mobile communication users worldwide today, according to GSM World[3]. In both markets, euphoria reached a critical peak between 1999 and beginning of 2000. Values on the stock market for telecommunication companies and internet-related businesses reached incredible heights. Against better knowledge, investors seemed to believe in unlimited growth, therefore bonds were traded at far too high prices, compared to the returns on investment.

[2] http://www.glreach.com/globstats/

[3] http://www.gsm.org/news/statistics/substats.shtml

These circumstances led to the effect that news about the next generation of mobile networks were greeted with euphoria. With UMTS, 3GPP had agreed on a new standard that would enable data rates of up to 2 Mbit/s over the radio interface. UMTS showed a possibility to merge mobile networks and the internet and therefore make mobile internet possible. The combination of the two fastest growing markets, internet and mobile communication, seemed promising to analysts as well as investors.

European mobile network providers payed a total of 130 billion Euro for UMTS licenses. In Germany and Great Britain, 50.8 billion and 38.5 billion Euro were payed, respectively. Although the licenses were sold at lower cost in other countries like Austria with 831.5 million Euro, one must not forget that most licensees are international companies involved in various European countries. Additionally, UMTS providers also have to pay for actually building UMTS networks. For Europe, these costs will sum up to another estimated 140 billion Euro [McKinsey02].

Stock exchanges took the high costs of investing in UMTS licenses extremely bad. On the other hand, because of the overall hype, the mobile operators were forced to compete. The dotcom bubble burst at the same time, and these effects led to massive losses on the stock exchanges, from which they still suffer today.

Even if income from speech telephony will account for some of the revenue in UMTS, the expenditures for UMTS can most likely be covered only by future data services like m-business or m-commerce.

It is common belief that only multi-media data services and interactive applications will generate enough traffic to justify UMTS networks. Therefore, video transmission and video telephony are thought to be the "killer-application"[4]. On the other hand, this only seems to be a means of "meaningfully" using the high data rates provided by UMTS. Transmission of moving images is not a new task: in 1880, Alexander Graham Bell developed the *Photophon*, a kind of optical telephone, where light was transmitted with mirrors. Transmission over longer distances and outside of buildings remained an unsolved problem. In 1927, Bell Labs for the first time managed to transmit moving pictures over

[4] Killer-application is jargon taken over from the computer industry, where it is used for an application program that intentionally or unintentionally gets consumer to bay the system the application runs on.

a telephone line from New York to Washington D.C. In 1936, Germany even established a television-telephone between Berlin and Leipzig, which was soon joined by other cities, but abandoned in 1940. AT&T developed the so-called *Picturephone* in 1970, which transmitted a new picture after several seconds. In a forecast, AT&T stated that by the end of the 1980s this service would become standard for every household and that by the end of the decade 85% of meetings would be carried out via picturephone or video conference. The 1990s again experienced a raise with the development of the ITU-T standard H.320 for desktop video conferences. This resulted in enhanced production of set top boxes and video phones. It was expected that falling prices, technological development, teleworking and cost reduction regarding business travel would further push the deployment of video telephony.

However, the history of video telephony and video conferencing shows that this business never managed to pass the phase of the hype. Later on, we will explain what can make a new service successful. However, it has to be kept in mind that the definition of a successful service is directly related to the number of subscribers. Only services that are used by many customers can be regarded as successful.

More important than video telephony are services based on localization of the mobile equipment, so-called *Location-Based Services* (LBS). The possibility of integrating the current geographical position of a mobile user into the application offered is one of the fundamental differences between stationary and mobile Internet. Therefore, Location-Based Services offer a wide range of new applications.

This realization led to a hype around LBS, which were now declared as not only one of the platforms for services in UMTS, but as the killer-application themselves. Analysts forecasted a total volume of $18 billion for the LBS market around 2000/2001:

- Strategy Analytics, December 2000: $16 billion in 2005.
- Ovum, January 2001: $20 billion in 2006.
- Analysys, February 2001: $2 billion by the end of 2002, $18.5 billion in 2006.

According to the forecasts stated above, there should have been an LBS market of at least 500 million dollar by the end of 2001. Although some mobile network providers offer some services, these have not yet reached a substantial volume. Not all providers have yet invested in necessary infrastructure, e.g., Gateway Mobile Location Center (GMLC). Also the attempt of the FCC (Federal Communication Commission) to localize all 911 emergency calls with a precision of 125 m have to be regarded as failed for the time being. A lot of the start up businesses financed by venture capital, which wanted to take part in the mobile Internet and LBS market, have already given up or changed their business models.

1.2 Making Services Successful

The history of telecommunications services in recent decades does not fill us with optimism that we know in advance what will success and what not. We cannot say that we really know what society wants, or even that we know how to go about finding an answer to this question. It is filled with market failures, like the Picturephone. Even the successes, like the Internet or SMS, were unexpected, and many of the applications of technologies, such as ISDN or ADSL, turned out to be different than the purpose for which they were designed.

In the telecommunications world there are two technology laws that affect every new solution - *Moore's law* of transistor scaling, and *Metcalfe's law* of user value scaling. Moore's law from the year 1965 originally states that chip density doubles every 12 months. G.E. Moore arranged his perceptions on a simple extrapolation of five data pairs (the number of integrable transistors over time in years), plotted on a semi-logarithmic sheet of paper. The data he used was collected by himself, over a period of six years. In the first time after publication of the results in [Moore65], attention to his findings was not so great. In 1975, Moore corrected his own forecast by setting his prediction of doubling period onto 18 months. Shortly after, his observation became generally known and accepted in professional circles. In 1995 Moore corrected his observation for the second time, now defining the doubling period with 24 months. It is generally agreed that the forecast will be valid as long the dimension of the smallest conductible structure on the carrying material does not penetrate into the atomic region. This is expected to happen in 2010 [Klussmann01].

Moore's law guarantees that technologies become obsolete and that economics become overturned at a rate that is incompatible with most infrastructure planning and financing. Metcalfe's law, on the other hand, is based on the observation that the value of a network grows with the square of the number of users. Every new user brings additional value to every one else using the service or technology. The overall value of the network to an individual user is small if few other users share that same network or application, whereas it becomes very large if many users are connected. The best example for this claim is the mobile telephony or the telephony itself. Thus, in the adoption of a new networked service there are two discernible regimes - a sparse regime in which there is little value to anyone and little incentive for new users to join, and a dense regime where there is great value for everyone and large incentive for new users. The huge barrier for any new service is getting from the first regime to the second. It is a barrier that few services have been able to cross.

G.A.Moore[5] [Moore02] responded to the question of technology adaptation by developing a general model. Moore's model of technology adaptation has gained much respect in the industry. He describes a cycle of technology adoption which corresponds to a Gaussian curve, if applied over a time axis (Figure 1.1).

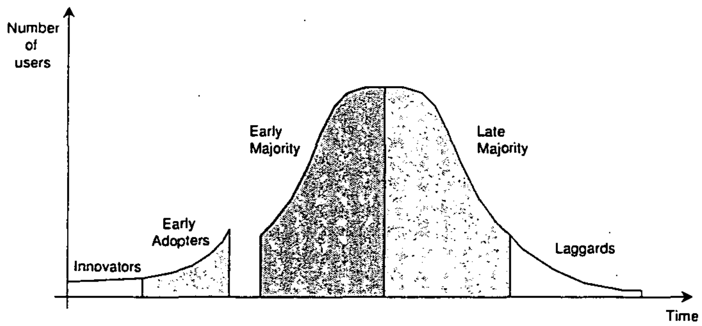

Figure 1.1: Revised technology adoption life cycle

In the beginning, the technology enthusiasts (the so-called *Early Adopters*) are in the center of interest. These customers are willing to spend more money on a new technology than the average customer. However, they are only a small number. In the group of *Early Majority*, a lot more customers can (and must) be reached. It depends from this measure, whether a new technology or a new

[5]Not the same Moore Moore's law originated from!

service will be successful or not. If this group can be reached, also more conservative customers and late comers will use the service, because technologies that are widely employed gain attractivity in many respects (Metcalf's law). Moore's model is especially interesting with regard to the fact that according to his findings there is a break in the market development, which occurs in the transformation from the Early Adopter to the Early Majority. This is the point, where high-tech businesses often fail, sometimes this is even true for a whole industry. Due to Moore, this effect is based on the totally different motives and behaviour of Early Adopters and Early Majority, respectively. Early Adopters are characterised by enthusiasm and visions with regard to new technologies, whereas members of the Early Majority tend to be more pragmatic and rational. Arguments that might convince the first group could prove to be of no importance to the second. Cheaper tariff models or improved product quality alone usually are not sufficient for success in a mainstream market. Even if a technology is very popular with the Early Adopters, it will not at all count for the Early Majority, as market rules for these two groups are total different [Ladstätter02].

G.A. Moore has described the model of technology adoption from the viewpoint of individual business strategies. He also explains why some companies are successful within the high-tech industry, while others, which may even offer the better products fail. This model can also easily be applied to technological developments like Location-Based Services, which have not yet reached the mainstream market. The industries involved can now take separate roads.

Today the hype around Location-Based Services seems to have passed the exaggeration phase - in the positive as well as in the negative sense. For those interested in LBS, like the telecommunications industry, the IT industry or geo-information services, the question arises how to transform a fascinating and promising technology into a economically sound market or, how G.A. Moore would formulate it - how to cross the chasm.

1.2.1 Methods for Reaching the Wide Market

It is obvious that the LBS market faces the same ups and downs as the m-commerce market or the previously mentioned market for video telephony. Meanwhile, such heavy market fluctuation is regarded as typical for the introduction of new technologies. The negative as well as the positive exaggeration of this phenomenon is nowadays referred to as *hype* (see Figure 1.2). A phase of positive overstatement and consequently excessive expectance is followed by disillusionment, disappointment and negative understatement, which finally gives way to realistic expectations. A sound growing market can only be achieved after these hype phases have been overcome.

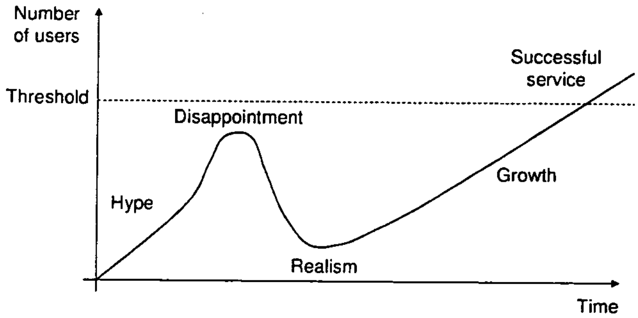

Figure 1.2: Technology or service hype

Picturephone was the most celebrated of a number of casualties of this startup problem. The initial users were asked to pay about \$ 100 per month for a service that had almost no value, since there was virtually no one who could receive a video call. A mathematical model that was used at the time to predict the market behavior was based on that used for the spread of a disease. The probability of an individual being exposed to a disease grows with the number of people who have contracted the disease. Thus the incidence of the disease shows an exponential increase with time, starting quite slow and then growing sharply before finally saturating. In the actual introduction of Picturephone only the first phase of this growth was observed, as the economics made it impossible to wait long enough to reach any critical threshold of acceptance. According to Figure 1.2, we could state that Picturephone came not out of the negative exaggeration after the hype phase.

Lucky [Lucky97] defines the critical threshold of acceptance (see Figure 1.2) as the border between the *sparse regime* with little value for users using the service and little stimulation for new users to join and the *dense regime* with high value for everyone and much stimulation for joining.

According to Lucky, there are three possibilities to reach the threshold. *Firstly*, the traditional way of concentration on a certain set of customers with high willingness to pay for services, usually private businesses, and to slowly migrate to wider customer numbers in the course of time. This was the case with mobile phones and facsimile machines. Even though it seemed as if the fax burst like a meteor on the horizon during the early 1980s, it had actually been invented more than a quarter of a century earlier. An important ingredient here was the agreement upon an international standard for facsimile transmission, and the subsequent embracing of this standard by the industry.

Secondly, offering services free or almost free of charge, as it was the case with the first Internet browser tool *Mosaic*. Before the National Center for Supercomputing Applications (NCSA) at the University of Illinois had offered the Mosaic browser for free download, personal computers had already penetrated into homes throughout the world, and increasingly they were equipped with modems for interconnectivity. The Internet itself had grown steadily for several decades, and was just reaching a critical threshold of user value based on email and ftp. Moreover, standards for document description and transfer, HTML and HTTP, had been adopted. Mosaic provided a simple user interface, it was for free, and could be easily downloaded anywhere in the world. Granted, without any pre-existing material to browse, there was almost no value in having a browser, but the price was right. Thus, the way out of the sparse regime of low user value was having a low user price. As soon as a few web sites started putting material on the net, and were suitably advertised by NCSA, the browser that had been downloaded mostly for reasons of curiosity started having incremental value. Suddenly the value was looking much greater than the price, and the explosion occurred.

The *third method* of passing the threshold is by offering governmental mandate and subsidy, as shown by the French Minitel system and the American National Science Foundation in the early days of Internet or the Korean government by supporting broadband penetration today. Lucky describes this as the *Field of Dreams* approach, which codifies a philosophy of service introduction, and represents the core of an old argument. Do services have to be known and quantified before an investment in infrastructure, or will revenue-producing service self-materialize when society is enabled with suitable infrastructure? "If we build it, they will come", was the famous reprise from the movie Field of Dreams. Engineers often invoke the Field of Dreams scenario, while accountants demand business cases and market analysis. Referring to the actual situation of UMTS and new services the question now is if the services will grow to fill the bandwidth? When this will happen? These questions are not merely philosophical, but serious matters of national economics. The dilemma for the network provider is whether the enormous investments in UMTS licences can be justified on the basis of return. Unforeseen technological alternatives could result in large stranded investment - the great fear of all carriers. It is hard to blame corporate executives and national policy makers who fear to rely on the Field of Dreams proposition. However, there is a catch in such arguments, because the Field of Dreams proposition also has a corollary - "If we do not build it, they surely can't come". If no one invests in new technologies and services, then surely we will not make new revenue streams. Moreover, we would probably not know what we were missing.

In the early days of the Internet, the American National Science Foundation used the Field of Dreams approach in the aggressive deployment of over-capacity in the network. This over-capacity enabled experimentation, and stimulated the growth of new services. The general opinion is that this approach is no longer possible in the commercial backbone of today, but the US government is currently attempting a reprise of the earlier success with a program called *Next Generation Internet* (NGI), with a capacity of 100 to 1000 times that of today's Internet.

Another very important measure for market development is *industrial standards.* They are a high importance

- If different industries have to cooperate,
- If none of the companies is strong enough to make its own standards, and
- To win the markets confidence.

All this is true for LBS. Meanwhile, standardization has been integrated as a means of market development by many companies in the IT business [Cargill01]. For Location-Based Services, two more or less complementary consortia have been established: the OpenLS Initiative of the OGC (Open GIS Consortium) and the Open Mobile Alliance (OMA). At the same time, a number of further groups are dealing with topics related to LBS, like 3GPP, the WAP-Forum, the IETF (Internet Engineering Task Force), the Bluetooth Special Interest Group, the MAGIC Services Forum, and the Open Group.

It is important to note that none of presented approaches can guarantee the success of a new technology or a new service. In general, it is not possible to forecast acceptance of a new technology or service within society, as this is mostly decided by social dynamics and dependent upon complex changes in social settings. In other words, we do not know what we want until we are told by the society. This leads to doubts regarding opinion polls and focus groups. Even experts cannot offer conclusive forecasts, because only the average person might be able to predict what he or she would want to have or do.

However, applications that help to save money, time and effort, have always been asked for. All these goals can be achieved by the employment of LBS services. In conjunction with mobility and wide-spread accessibility (even indoors), the functions of LBS exhibit massive potential for future data flows and consequently offer new revenue sources for mobile network providers.

1.3 Classification of Location Based Services

With the classification of location-based services one can get a better impression what positioning is capable of. Thereby it should be said that the presented classification does not set any finite frame. The classes can be extended at will. Especially the given examples should be understood only as such. Quite soon it becomes clear, that the number of possible applications could be really huge. How huge depends on several factors.

The most important factor is the art of control over the LBS platform. There are three possible scenarios. The first scenario would be that the network provider has the full control over his platform, being the only LBS service provider. Nowadays this is the prevalent model in Europe and USA. The number of services depends strongly upon how many resources the network provider allocates for the deployment. Significantly more services are possible with the second scenario, where the LBS platform is open for third party service providers. A good example is the i-mode service of the biggest Japanese mobile network provider NTT DoCoMo. Third party service providers are able to offer their services to customer of the network provider by using the network provider's LBS platform. By doing so, the number of services is much bigger. Particularly, if using such a good business model as NTT DoCoMo did. The third scenario is a complete open platform. The best example for this is the Internet. Network providers allow the use of their resources for a given amount and the customer is free to use it how he would like to. Thereby an incredible amount of services would be generated, which in turn creates new needs, leading to permanent enhancements.

Beside the usage scenarios of the LBS platform, a further factor influencing the deployment of location-based services is the positioning accuracy. The higher the accuracy, more applications are possible. We will come to talk about this in Section 2.7. There are also other, non-technical issues, on how to reach the wide market, discussed in the previous section. Last but not least the success also depends on the usability of the services. However, the KISS principle ("Keep it simple and stupid") should be the premise of all product developments.

According to the GSM association, there are three basic types of location-based services [Association03]:

- Pull,
- Push, and
- Tracking.

Pull services are services where customers make a request for location-based services by themselves. When making the request they also give permission for their position, because without that location information the request for service cannot be completed.

Push services differ from pull services on the point that the request for service is not technically made by the customer, but by the service provider. In this case the customer must give permission for the service provider to send information to his mobile phone.

Tracking services are services where someone (person or service) asks for a location of the mobile terminal (person, vehicle, fleet, etc.). As in the pull and push cases, the assumption is that the customer has given the permissions which allow particular persons or services to track him.

Table 1.1 shows a few LBS categories and exemplary applications. Usually it is clear of what type the application is (push, pull, or tracking), but it can also depend on the art of implementation.

Categories	Service	Description
Emergency and Assistance Services	E911 and E112	Enabling mobile or cellular phones to process 911 (in the USA) or 112 (in Europe) emergency calls and enable emergency services to locate the geographic position of the caller.
	Roadside Assistance	In the case of having a car accident, running out of gas, having a flat tire, or needing a tow, the nearest service point will be provided with position information of the car.
	Medical Alerts	Providing position if lost during hiking, injured on skiing tours, being unable to speak during an emergency call, or some other situation where life could be in danger.
Navigation	Direction and Destination Guide	The classical navigation to predefined destinations.
	Traffic Management	Specific action, according to the amount of traffic in some geographical region.
	Indoor Routing	Navigation guide inside buildings.
Information	Travel Service	City guide with lot of interesting information about buildings passing by.
	Mobile Yellow Page	Navigation to points of interest like cafes, restaurants, theaters, shops, museums, etc.
	Infotainment Service	What's up in the theater passing by?
Advertising	Banners	LBS applications of any kind are given for free. The use of these applications is financed by banner adverts.
	Alerts	Lunch time alert, if passing nearby my favorite kind of restaurant during a predefined lunchtime window.
	Advertisements	Shops are advertising their goods or giving special discount to passers-by.
Monitoring	Fleet Tracking	Tracking, maintaining, and otherwise managing a fleet of vehicles based on positioning information.
	Pet Tracking	Tracking of cats, dogs and other domestic animals.
	Tracking Games	Playing life adventure or action games with other, position aware teammates.
Billing	Location Sensitive Billing	If making a phone call from home, the tariff is different than making the call from outside the "home-zone".

Table 1.1: Classification of location-based services

1.4 Technological Evaluation

As we have seen in Section 1.2, success cannot be guaranteed for services that are in the hype phase because various factors decide about success or failure. Left alone that LBS does fulfil some basic prerequisites for a successful service, there still are technological aspects that have to be considered. In the next step, it is necessary to realize which technical components will be vital for a breakthrough of LBS. Here we will have to talk about data rates, positioning methods, terminals, architectures and last but not least the transmission Quality-of-Service (QoS).

1.4.1 Data Rates

Today's GSM networks offer a data rate of 9.6 kbit/s[6] and medium speech quality. In comparison, ISDN offers data rates of 64 kbit/s and good speech quality. In contrast to speech transmission, data transmission is feasible with any data rate, however, the requirements for reliability are much more demanding. For customer acceptance and interactivity features, data rates again play a more important role. If we take the size of a web page to be 20 Kbytes, then download time with GSM will amount to 16 seconds. This will not be tolerated by users.

In this respect, GPRS is more interesting, which provides realistic data rates of about 43 kbit/s. Here, download time for a 20 Kbytes web page will amount to 3.7 seconds. For web pages specifically tailored for mobile equipment, e.g., with Windows CE and even smaller size than usual internet pages, performance is acceptable. What makes GPRS even more appealing is the fact that because of its packet-oriented transmission scheme, cost is charged according to the transfered volume. Today almost all European network providers offer GPRS. UMTS promises a bandwidth of 2 Mbit/s, but these data rates are only provided for pico-cells in the UTRA TDD mode and the 2 Mbit/s have to be shared among all active users in this cell. The usual data rate in UTRA-FDD mode will be 384 kbit/s, being about six times as much as for ISDN, is still a very attractive measure compared with today's developments.

The new trend is also going in the direction of Wireless Local Area Networks

[6] Small "k" stands for 1000 and "K" for 1024.

(WLAN and further developments like WiMax). At the beginning, these were seen as a possible competition. Now they are viewed as an opportunity for widening the bandwidth on hot spots to 10 and more Mbit/s [7].

1.4.2 Positioning

Several methods exist for locating mobile terminals in today's mobile networks. These methods can be discerned by their accuracy of location or by the cost of overall installation, respectively. Basically, all methods can be divided into three groups (Figure 1.3). The first is based on available addresses, signalization and measurements in existing mobile networks (Cell-ID[8]) and Network Measurement Results (NMR) like Timing Advance (TA) and signal strength measurements. The second makes use of triangulation, with a differentiation between network based (*Uplink Time-of-Arrival* (UL-TOA) and *Angle of Arrival* (AOA)) and equipment based methods (*Enhanced Observed Time Difference* (E-OTD) and *Observed Time Difference of Arrival* (OTDOA)). The last group additionally uses satellites to locate the exact position (*Assisted Global Positioning System*, A-GPS).

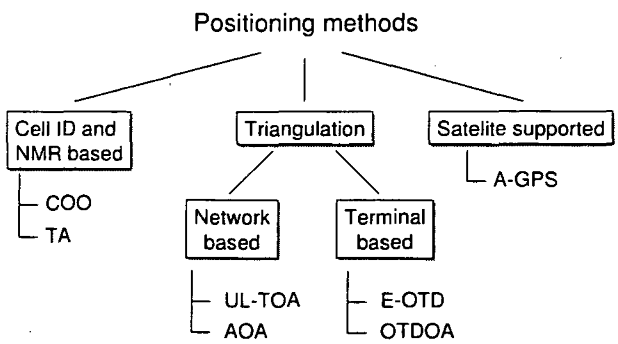

Figure 1.3: Positioning methods

Hybrid location techniques combine several of the methods described above to provide positioning estimates with better accuracy, reliability and coverage, including indoor, outdoor, urban and rural areas.

[7] 802.11b with 11 Mbit/s, 802.11g with 54 Mbit/s, and WiMax up to 75 Mbit/s.

[8] The stand alone version of the Cell-ID method is often termed as the *Cell of Origin* (COO) method.

Together with specialized UMTS procedures like IPDL (Idle Period Downlink) and newer developments, these classic and hybrid methods of location are described in more detail in Chapter 2. The variety of possible applications is closely related to the accuracy of location, as a wider range of applications is feasible with a more exact location.

1.4.3 Terminals

The availability of appropriate terminals is essential for providing customers with a certain service. Concerning LBS, the end device depends on the method applied. For the Cell-ID method, no modifications have to be applied to terminals and all existing mobile equipment can be employed. This is the big advantage of this method, however, the drawbacks are the relatively small displays and more or less non-existent software platforms for today's mobile terminals. Moreover, the Cell-ID method is a comparatively inaccurate method, which excludes a variety of possible applications.

Nowadays, the trend for mobile terminals is clearly in the direction of smartphones and mobile digital assistants. This means that these new terminals cannot only be used for telephoning, but also offer an organizer with additional features like MP3-player, game console, digital camera, etc. This also includes additional software and not only WAP browsers, but XML, Java Virtual Machine or even a certain operating system like Symbian, Windows CE, Palm, or Linux. It will therefore be possible to run a wide variety of different applications on a next generation mobile end device. Not only processor power will offer more performance, but also the displays of these new devices will be bigger and look more like those of today's Personal Digital Assistants (PDAs). Combining this facts with the new trend of integrating new radio interfaces like WLAN, Bluetooth or RFID[9] into this devices, the future of LBS becomes even brighter. We are also positive that new problems that will certainly arise from these developments, e.g., higher power consumption, can be solved.

Next generation terminals will be able to employ methods like E-OTD (Enhanced Observed Time Difference). Concerns that the size of the terminals could be more than what we are used to today, are completely unsubstantiated. This can be illustrated by the history of GPS receivers. In 1995, one of

[9]Radio Frequency Identification (RFID) is a method of remotely storing and retrieving data using devices called RFID tags.

the leading companies in embedded GPS systems, Sigtec Pty Ltd.[10], manufactured a GPS receiver of 125 x 100 mm^2 (Signav G4320). The current model Signav MG5003 measures 28 x 28 mm^2 and the future model (microNAV, available in 1Q05) will only be 14 x 13 mm^2. Also Benefon's[11] example shows that their Benefon Esc! NT2002, which is a GSM phone and a GPS receiver in one, weighing 198 g is not bigger than a usual GSM mobile phone.

A bigger display can make use of several navigation applications. In contrast to conventional GPS systems, a mobile network provider can offer much better receiving quality in urban areas and indoors, especially with E-OTD, AOA (Angle of Arrival), or UL-TOA (Uplink Time-of-Arrival), but also with A-GPS (Assisted Global Positioning System). Moreover, it can be seen that today's fixed car navigation systems will be replaced by so-called off-board mobile devices. This can be compared to the change from analog fixed car phones to today's digital mobile phones.

1.4.4 Network Architecture

Additional hard- and software have to be implemented in order to employ LBS in mobile networks. In general, mobile networks can be divided into three parts: the *access network* (BSS, Base Station Subsystem) with base stations and controller, the *core network* (NSS, Network Subsystem) with switching nodes and data base, and the *backbone* for connection of the switching nodes (Figure 1.4).

New hardware that will have to be implemented consists mostly of a new network node within the core network, called *Mobile Location Center* (MLC). The MLC is responsible for a set of tasks such as privacy, authorisation and authentication, delivery of location information to authorised applications, billing and charging, access to base station co-ordinates, and other physical parameters required for location. The MLC can further be divided into a *Serving Mobile Location Center* (SMLC) and a *Gateway Mobile Location Center* (GMLC). The Gateway-MLC works as access node for requests from LBS applications and executes authorization.

Positioning requests are transferred to the current MSC (Mobile Switching

[10] http://www.signav.com.au
[11] http://www.benefon.com/

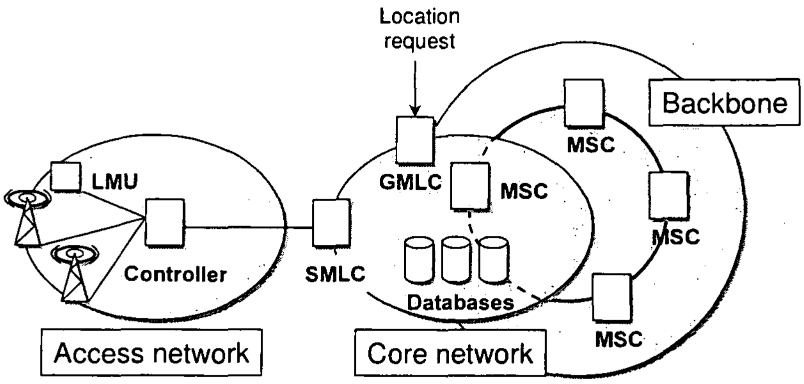

Figure 1.4: General LBS architecture

Center), which is also connected to the Serving-MLC that actually performs the positioning within the mobile network. Depending on the positioning method and its implementation, the whole calculation of the position can also be carried out by the Serving-MLC. Furthermore, a *Location Measurement Unit* (LMU) can exist in the access network, which then works as a reference point for positioning methods like E-OTD and UL-TOA. If necessary, also the Base Station Subsystem should support the applied positioning method. Of course, all of these network nodes have to be interconnected with each other and with the existing architecture. Definition of new interfaces will therefore be necessary. Particulary, standardization will play a decisive role for the success of LBS. Chapter 3 will discuss the topics of new interfaces for GSM, GPRS, and UMTS networks as well as specifications of LBS architecture.

1.4.5 Transmission Quality-of-Service

Another important factor for the success of a new service will be its quality. There are three notations of Quality-of-Service (QoS) defined in [Gozdecki03] - intrinsic, perceived, and assessed.

Intrinsic QoS pertains to service features stemming from technical aspects. Thus, intrinsic quality is determined by transport network design and provisioning of network access, terminations, and connections. The required quality is achieved, among other things, by an appropriate selection of transport protocols, QoS assurance mechanisms, and related values of parameters. Intrinsic QoS is evaluated by comparing measured and expected performance characteristics. User perception of the service does not influence the intrinsic QoS rating.

Perceived QoS reflects the customer's experience of using a particular service. It is influenced by the customer's expectations compared to observed service performance. In turn, personal expectations are usually affected by the customer's experience with a similar telecommunications service and other customer's opinions. Thus, the QoS with the same intrinsic features may be perceived differently by various customers. It follows that just ensuring particular service (network) parameters may not be sufficient to satisfy customers who are not concerned with how a service is provided. The QoS offered by a provider must reflect the intrinsic QoS as well as some non-technical parameters that are meaningful to the customer and relevant to a particular community's expectations.

Assessed QoS starts to be seen when the customer decides whether to continue using the service or not. This decision depends on technical and non-technical reasons discussed in Section 1.2.

Assurance of a satisfactory level of intrinsic, perceived, and assessed QoS may be considered separately. The main focus of this work is intrinsic QoS. Part of intrinsic QoS could also be qualitative measures like scalability, accessibility, feasibility, robustness, etc. but we are more focused on quantitative measures or measurable values. These can be understood as technology-dependent performance measures that have to be provided by the network in order to insure appropriate transmission quality for a certain service. These parameters comprise delay, jitter, and data loss. It can be seen clearly that these measures will play a decisive role for real-time services but these holds also for the perceived QoS of LBS services. A service with poor performance can hardly be successful.

In traditional telephone systems (PSTN, Public Switched Telephone Network), circuit switching technology established a connection between the calling user (usually termed user A) and the called one (user B) before the conversation can take place[12]. If the conversation is a data session, all data for this connection

[12] The "conversation" is often referred to as POTS, Plain Old Telephone Service, meaning telephone service or calling without any service features.

are routed over the same transmission channel, which is exclusively reserved for this connection. Therefore, delay for all packets is the same and no jitter occurs. The order of the packets is maintained and the probability of packet loss is very low. This results in very high QoS but the drawback, however, is that the path is exclusively reserved for this connection over the whole period of time. Even when user A or B do not transmit any data, transmission capacity cannot be made available to other users. Conversation between humans is characterized by the fact that only 40% of the time of a dialog is used for phonetic performance by one of the speakers. In an average telephone call, the line remains unused in 20% of the time [Klussmann01]. This waste of capacity leads to under utilization of the network and should be avoided.

In packet switched networks, a connection is not provided with a dedicated physical channel. The sender segments data into single smaller or larger packets, and additional data for identification are added in a header. These packets are routed through the network independently. Efficient use of network resources is achieved, because the network is only loaded if transmission of data is in course. This also makes more flexible resource allocation possible, whereas in circuit switched networks only whole-numbered multiples of a transmission channel capacity can be allocated[13]. However, packets of the same connection can take different routes through the network and experience different delay, which then causes jitter. Even the correct order of packets can be lost, or the network can experience overload, which again leads to packet loss. Therefore, QoS guarantees cannot be given in solely packet switched networks. A part of this thesis deals with the problem of knowing where, how, and which QoS mechanisms should be employed in order to realize new applications like Location-Based Services but also support existing real-time transmission of voice. Chapter 4 will show QoS architectures in more detail.

1.5 Basic QoS Terms within the Internet Protocol

In the previous section we have seen that QoS parameters such as throughput, end-to-end delay, jitter, and packet loss ratio, cannot be guaranteed in packet switched networks. At this point, one should note that this is only partly true. Existing packet switched networks as ATM, X.25, and Frame Relay

[13]e.g., digital speech channels at 64 kbit/s.

(FR) are implying mechanisms for QoS provisioning. The problems lie in IP based networks - they do not. Because of the wide spread of IP based networks and IP based applications, the IP protocol stack is also applied to the GPRS and the UMTS architecture. Based on the best effort service model provided by the IP protocol, it is important to analyze mechanisms like congestion, fairness, flow control, feedback, scheduling, and queuing management. These mechanisms are necessary for the elaboration of QoS architectures.

1.5.1 Congestion

Congestion is the state of sustained network overload where the demand for network resources is close to or exceeds capacity. IP networks have suffered from the problem of congestion, which is inherent in best effort packet networks due to uncoordinated resource sharing. It is possible for several IP packets to arrive at the router simultaneously, needing to be forwarded on the same output link. Clearly, not all of them can be forwarded simultaneously; there must be a service order. Buffer space must be provided as temporary buffer for the packets still awaiting transmission, after the routing decision. Buffering incoming packets, making the routing decision, and forwarding the packet are the main tasks of each router (Figure 1.5).

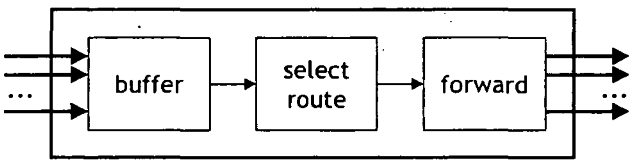

Figure 1.5: Router tasks

Sources that transmit simultaneously can create a demand for network resources higher than the network can handle at a certain link. Buffer space in the routers offers a first level of protection against a sudden increase in traffic arrival rate. However, if the situation persists, the buffer space is exhausted and the router has to start dropping packets. Traditionally, routers have used the first-come first-serve (FCFS) service order, typically implemented by a first-in first-out (FIFO) queue, and drop from the tail upon buffer overflow as their queue management strategy. The problem of congestion cannot be solved by introducing "infinite" buffer space inside the network; the queues would then grow without bound, and the end-to-end delay would increase. Moreover, when packet lifetime is finite, the packets leaving the router would have timed out already and been retransmitted by the transport protocols [Nagle87].

In fact, too much buffer space in the routers can be more harmful than too little, because the packets will have to be dropped only after they have consumed valuable network resources.

Congestion can cause high packet loss rates, increased delays, and can even break the whole system by causing a congestion collapse. This is a state, where any increase in offered load leads to a decrease in the useful work done by the network (Figure 1.6, the dotted curve).

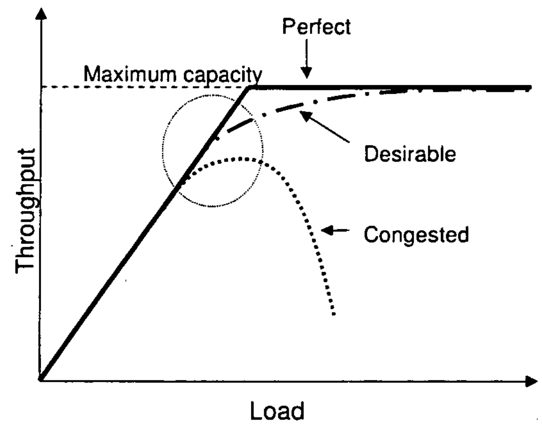

Figure 1.6: Overload control

The threat of a congestion collapse is not new. It dates back to the early days of the Internet (then ARPANET) and can take several forms. In 1984, Nagle [Nagle84] reported on a congestion collapse due to TCP connections unnecessarily retransmitting packets that either were in transit or already received at the receiver. This form of congestion collapse is a stable condition, which results in throughput that is only a small fraction of the normal throughput. This phenomenon, as predicted by Nagle, occurred several times during 1986 – 1987 with a large number of sites experiencing simultaneous slowdown of their network services for prolonged periods [Gevros01]. At that time BBN

Technologies[14] , the company maintaining the Internet backbone, responded to the collapse by providing additional link capacity (overprovisioning).

This form of solving congestion problems is seen by some as a temporary fix but on the other side there are new technologies like DWDM (Dense Wavelength Division Multiplexing) available, providing capacity extensions of the backbone by only activating more optical wavelengths on the same fibre cable. The advantage of overprovisioning is the simplicity but the main disadvantage that it usually cannot be realized end-to-end. The scarce resources at the radio interface can be seen as the bottleneck for the end-to-end realization in wireless networks and in this context also the cost of overprovisioning, but on the other hand also QoS mechanisms do not come for free. More about overprovisioning will be given in Chapter 7.

Another form of congestion collapse is the one coming from undelivered packets. In this case, bandwidth is wasted by delivering packets that will be dropped before they reach their final destinations. Compared to the "classical" congestion collapse reported by Nagle, this form of congestion collapse is not a stable condition but one that can be reversed if the offered load is reduced. Other forms of congestion collapse, reported by Floyd [Floyd99], include the fragmentation-based congestion collapse and the congestion collapse from exhausted packets. The fragmentation-based congestion collapse occurs when the network transmits fragments of packets that will be discarded at the receiver since they cannot be reassembled into a valid packet. The congestion collapse from exhausted packets occurs when the network carries packets that are no longer wanted by the user (because the transfer took so long for instance).

In order to deal with congestion, the Internet used end-to-end window-based flow control in its *Transmission Control Protocol* (TCP) [Postel81], primarily for controlling demand on the receiver's bottleneck resources (memory and processing). Since 1987, TCP congestion control has been augmented with the Slow Start and Congestion Avoidance algorithms developed by Jacobson [Jacobson88]. These algorithms became mandatory requirements for all Internet hosts [Braden89]. The receiver-driven TCP flow control mechanisms have been the only congestion control methods available. This is why the term *flow*

[14]BBN Technologies, originally called "Bolt, Beranek and Newmann", is a technology company that provides research and development service. They are based in Cambridge (MA, USA) and are also a defense contractor.

control is sometimes confused with *congestion control* although the former is only one method of the latter.

1.5.2 Fairness

Fairness is conceptually related to congestion control. Under conditions of low load, everybody's demands are satisfied and there is no need for trade-offs and no considerations for decisions that lead to fair allocation of resources. Fairness becomes an issue only when there are unsatisfied demands and users have to compete for their share. In an environment of competitive individual users, the critical factor of cooperation relies on the underlying notion of fairness as well as incentives for adopting certain behaviors. The importance of fairness is a result of an optimization under uncertainty argument.

Although several definitions of fairness arise from various disciplines, in the networking world the most popular notion is indeed that of *max-min fairness* (also referred to as the classical notion of fairness). Max-min fairness is informally defined as "each user's throughput is at least as large as that of all other users which have the same bottleneck" [Jaffe81]. Fairness can be examined macroscopically along a path (global view) or on a per-link basis (local view). To translate from a locally fair allocation decision to a globally fair one, each user (e.g., flow[15]) should limit its resource usage to the smallest locally fair allocation along its path. This is known to result in a globally fair allocation [Bertsekas92].

Max-min fairness is a widely used technique for resource allocation in cases where some users demand is smaller than others. It has been considered desirable in the networking community - in both the Internet Engineering Task Force (IETF) and the ATM Forum - and operates as follows:

- Resources are allocated in order of increasing demand,
- No source gets a resource share larger than its demand, and
- Sources with unsatisfied demands get an equal share of the resource.

According to [Keshav97], this formal definition corresponds to the following operational definition. Consider a set of sources

[15] A flow defines a stream of packets with the same source and destination addresses and port numbers.

$$1, .., n$$

that have resource demands

$$x_1, x_2, ..., x_n.$$

Without loss of generality, order the source demands so that

$$x_1 \leqslant x_2 ... \leqslant x_n.$$

Let the server have a capacity C. Then, we initially give

$$C/n$$

of the resource to the source with the smallest demand, x_1. This may be more than what source 1 wants, so that

$$C/n - x_1$$

of the resource is still available as unused excess. We distribute this excess evenly to the remaining $n - 1$ sources, so that each of them gets

$$C/n + (C/n - x_1)/(n - 1) \tag{1.1}$$

of the remaining capacity. This process ends when each source gets no more than what it asks for, and, if its demand was not satisfied, no less than what any other source with a higher index obtained.

As we have seen, initially all users get at least as much as the "smallest" user demands, and the remaining resources are evenly distributed among the users with unsatisfied demands. It follows that from those users with unsatisfied demands no one can increase its share without decreasing the share of a user with an already small one.

This can be formally expressed as follows: a vector of allocations x is max-min fair if for any other feasible vector y there exists a user j such that

$$y_j > x_j$$

implies that there exists user i such that

$$y_i < x_i < x_j.$$

On the other hand, fairness should not necessarily imply equal distribution of resources to all those users with unsatisfied demands. A fair allocation of resources is usually defined with respect to a given *policy*. Policy is the unified regulation of access to network resources and services based on administrative criteria. It can be expressed at different levels [Rajan99]:

- Network Level – taking into account topology, connectivity, end-to-end performance objectives and the dynamic state of the network.
- Node Level – where a set of mechanisms like classification, policing, buffer management, and, scheduling allow administrative intentions to be translated into differential packet treatment.

Under a certain policy, it may be justifiable to allocate more resources to some users than to others. This leads to a generalization of max-min fairness. *Weighted max-min fairness* generalizes the concept of max-min fairness for the case where users have different rights to resource allocation. Each user i is associated with a weight w_i, which reflects its right to a relative resource share. The weighted max-min fair allocations are calculated as follows:

- Resources are allocated in order of increasing demand normalized by weight,
- No source gets a resource share larger than its demand, and
- Sources with unsatisfied demands are allocated shares in proportion to their weights.

The formal definition is similar to the one used for max-min fairness, but the allocations are replaced by the ratio x_i/w_i. It has been argued that max-min fairness can be suboptimal in several contexts depending on the actual utility functions of the flows. The utility function describes the utilization proportion of the flow on the entire load of the output link.

Using logarithmic utility functions Kelly introduced the notion of proportional fairness [Kelly97] as a more suitable fairness model for bandwidth sharing. Proportional fairness tends to favor short flows over longer ones.

1.5.3 Proportional Fairness

Formally, a vector x is proportionally fair if it is feasible and if for any other feasible vector y the aggregate of proportional changes is zero or negative:

$$\sum_i \frac{y_i - x_i}{x_i} \leq 0. \tag{1.2}$$

In a similar fashion, weighted proportional fairness generalizes the notion of proportional fairness for the case where user allocations are influenced by the price per unit share a user is prepared to pay (w_i). Then the feasible vector of allocations x is weighted proportionally fair if for any other allocations vector y the weighted sum of the proportional changes is zero or negative:

$$\sum_i w_i \frac{y_i - x_i}{x_i} \leq 0. \tag{1.3}$$

The allocations resulting from the two types of fairness described above are illustrated with an example. Figure 1.7 shows three flows sharing two links of unit capacity. Let the notation for the allocations to flows 1, 2, and 3 be

$$x_1,\ x_2,\ x_3$$

then the max-min fair allocations would be

$$1/2,\ 1/2,\ 1/2$$

while the proportionally fair allocations would be

$$1/3,\ 2/3,\ 2/3.$$

Assuming a logarithmic utility function for each flow, proportional fairness requires that the longer flow 1 sacrifices its own utility (thus receiving a smaller share) for a greater sum of the utilities of all flows. It is known that TCP is biased toward shorter flows; flows that traverse a smaller number of links (because they are exposed to potentially fewer losses) or have smaller round-trip time (because they update their window faster). However, the fact that

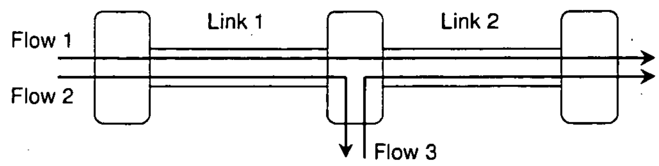

Figure 1.7: Flows sharing links of the same capacity

proportional fairness results in allocations that favor such short flows, does not necessarily mean that TCP is proportionally fair. Clearly, max-min and proportional fairness are the same in the case of a single resource. Proportional fair shares depend on the number as well as capacity of the resources (links) on which a user places demand. The problem with this approach is that the end user is usually unaware of how many or how congested the links are on which the traffic goes through. The user tends to view the network as a single resource, unaware of the actual cost implications different actions might have. These costs are reflected in the prices the user is asked to pay, but again it is hard for the endpoints to infer and the network to calculate these prices and communicate them to the endpoints.

1.5.4 Congestion Control Mechanisms

We identify two broad classes of congestion control mechanisms with regard to where these mechanisms are implemented:

- Host-based, and
- Router-based mechanisms.

In the Internet architecture, congestion control mechanisms are implemented in the end hosts by the TCP protocol. Upon detection of congestion the sources should send their packets into the network more slowly. This mechanism is called *end-to-end flow control.* In order for a host to be able to detect congestion, the routers must be able to provide the information that the network is currently (or is about to become) overloaded; this mechanism is called *feedback.* Flow control and feedback are conceptually related, so they are often referred to as *feedback flow control.* Although flow control should be aware of the feedback semantics, the exact mechanisms used to implement either are orthogonal.

The feedback mechanism is distributed and can be implemented partly or entirely at the end hosts or routers. Packet drops were the only means for a router to cope with congestion. The sources become aware of the packet drops, interpret them as a congestion indication, and reduce their rates. The feedback from the network and the response from the source are the foundations of Internet congestion control. They are very important because they facilitate decentralized resource allocation. However, with decisions made at the end hosts and treatment of the network as a black box that simply drops packets, there is clearly a limit on how much control can be achieved over the allocation of network resources. This also limits the range of services the network is capable of offering if using the TCP/IP protocol suite. Routers, on the other hand, exactly know how congested they are and can therefore perform more drastic resource management, as it is the case in ATM (Asynchronous Transfer Mode) networks. Thus, the introduction of router mechanisms for congestion control that will enable the network to more actively manage its own resources seems promising [Floyd99].

These mechanisms can be used as building blocks for providing higher-level resource management mechanisms such as link sharing, penalty boxes, and pricing, which by means of financial incentives controls the sharing of network resources. The extension of router functionality does not contradict the design philosophy of the Internet where all state should be kept at the edges of the network. Routers have two conceptually orthogonal methods of managing their own resources:

- Service discipline – to directly manage bandwidth allocation on an output link, and
- Queue/buffer management – to manage buffer space and queue occupancy, respectively, and thus indirectly affecting bandwidth allocation.

Together these two concepts are called the scheduling discipline of the router and a main part of this work is dedicated to this important topic.

Congestion can also be avoided at the expense of low resource utilization (overprovisioning). In Chapter 7 we will do some performance evaluation, showing how overprovisioning influences the network QoS. The goal of any congestion control mechanism, with respect to resource utilization, is to operate the resource (link) in a region close to its capacity (dot-dash line in Figure 1.6).

1.5.5 Flow Control

In control theory, a controller changes its input to a black box and observes the corresponding output. The goal is to choose the input as a function of the observed output so that the system state conforms to some desired objective, provided that the system state can be observed. From a control-theoretic viewpoint, the end host flow adjustment is the response to a servo-control loop, which needs to match the source sending rate to the rate that corresponds to its fair share at the bottleneck link.

The problem is that the appropriate bottleneck service rate becomes known to the source after a delay, and the new rate (after any adjustments) takes effect at the bottleneck only after another delay. The precision of the servo-control loop determines performance; if the queue at the bottleneck link is empty, throughput will be less than the maximum. If there are always packets in the queue, the link will never be idle, but if the queue size grows beyond a limit, packets will start to be discarded. However, in flow control, the output of the system does not depend only on the actions of that particular flow, but also on the actions of all other flows sharing the same path. Other flow control issues involve the *decision function* (how the feedback information is interpreted), *frequency of control* (how often the source decides to adjust its window/rate), and *control function* (how the window/rate is adjusted).

The above issues depend on the end-to-end network path properties (available bandwidth, delay, and loss) as well as the nature of the feedback signal, which has to be predictable and well defined; otherwise, end-host adjustments cannot be used to implement resource allocation policies or even to control loss. Except for the primary goal of rate matching, a flow control mechanism tries to achieve certain, sometimes mutually contradictory, objectives that allow interesting design trade-offs and lead to a wide range of mechanisms. Flow control schemes generally fall in two broad categories:

- Open loop, and
- Closed loop.

Open-loop flow control is acceptable only in an environment without considerations about the impact of individual actions to other network users. In an open-loop flow control scheme the sender describes its rate to the network with parameters like burst size and inter-burst interval. Simply stating the rate is not sufficient, because b *packets/s* may be one packet every $1/b$ seconds. It can also mean a burst of b back-to-back packets every second, which might be unacceptable for a router that does not have enough buffer space to buffer the burst. The network examines the parameters given by the sender. If the request can be granted (whereby the admission control is based on availability or policy criteria), it reserves resources (corresponding to these parameters) along the path from the sender to the receiver. The sender simply ensures that its rate conforms to the given description, and in this fashion, network congestion is avoided. This paradigm fits nicely in a connection-oriented QoS architecture like IntServ, but cannot be enforced with end-to-end mechanisms only. It requires resource management mechanisms in all the routers.

The difficulty with open-loop flow control is to accurately describe source behavior using a small set of parameters since the network must be aware of these parameters for admission control calculations. However, sources may be bursty and delay-intolerant; reservations at the peak rate do not usually lead to the most efficient use of bandwidth, preventing statistical multiplexing gains. It is therefore useful for the source output to be smooth. Other options for source description parameters include average rate or the use of a Linear Bounded Arrival Process (LBAP) [Keshav97]. An LBAP-constrained source bounds the number of bits transmitted in a time interval by a linear function of time:

$$A(t) = \rho t + \sigma, \tag{1.4}$$

where ρ is the long term rate allocated by the network to the source (which may be substantially larger than the source's true average rate), and σ is the longest burst a source may send. By this, LBAP can be used to describe a source with a known long-term average sending rate, which can occasionally deviate from that average and transmit in bursts of a known maximum size.

A *token bucket regulator* is a mechanism for regulating the size of bursts allowed to a source characterized by LBAP [Turner86]. Intuitively, the regulator collects tokens in a bucket and sends a packet only if the bucket has enough tokens otherwise the packet waits until enough tokens have been accumulated in the bucket or until it is discarded (Figure 6.13).

The effect of a token bucket is to limit the size of bursts to a little more than the buckets depth (since tokens can arrive while packets are being transmitted). More about token bucket will be given in Section 6.5.1.

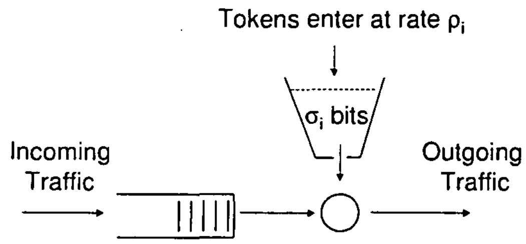

Figure 1.8: A token bucket

Closed-loop flow control schemes target more dynamic network environments, where it is a requirement for the sources to dynamically adapt their rate to match their fair share of network resources. The fair share usually fluctuates, and the sender must be able to track these changes and adjust its rate, in response to feedback signals, to allow for more efficient resource utilization. If closed-loop flow control is ineffective, sources either suffer excessive packet loss or underutilize network resources. The first generation of flow-control protocols did not explicitly consider the network state; they simply matched the source rate to the service rate at the destination. The three important protocols in this generation are *on-off*, *stop-and-wait*, and *static-window* flow control [Keshav97].

The second generation of protocols changes the source rate in response to both the receiver state and the network state. According to the choice of control, closed-loop schemes can be *dynamic window*, in which the source indirectly controls the transmission rate by modifying the number of packets sent but not yet acknowledged. It can also be a *dynamic rate*, in which the source, every time it sends a packet, sets a timer with a timeout value equal to the inverse of the appropriate transmission rate and transmits the next packet when the timer expires. The potential damage to the network is constrained in different ways, but window-based schemes are easier to implement because they do not require a fine-grained timer, which is hard to implement in non-real-time operating systems [Keshav97].

1.5.6 Feedback Mechanisms

The mechanism used for notifying the sender about network congestion or the appropriate sending rate is called *feedback*. It inherently involves both the routers that generate the congestion messages and the receiver host that forward the message to the sender for interpreting it accordingly. Closed-loop flow control mechanisms and overall network performance rely heavily on feedback. Without a feedback mechanism a source would be helpless as to what to do with its sending rate, and the network could become unstable, unfair, and either congested or under-utilized. Feedback involves information about the state of the system, so in principle it should originate from the network and ultimately be delivered to the sender. The sender receives feedback either directly from the network or from the network via the receiver. Therefore, there are two forms of feedback:

- implicit, and
- explicit feedback.

Implicit feedback requires the end-hosts to be responsible for monitoring the performance of their own transmissions (delay, loss) for indications that will let them infer the state of the network and determine their appropriate sending rate. Nevertheless, it is debatable how accurately this can be derived. The most common form of implicit feedback is packet drop and has been traditionally used by Internet routers. However, packet drop is not necessarily an indication of congestion.

The best example is the error prone wireless link, which has been very well described and evaluated by Taferner [Taferner02]. Another proposed method of implicit feedback is the observation of the rate at which packets emerge from the bottleneck [Keshav91] or the measurement of the change in end-to-end delay as the transmission rate changes [Brakmo94]. The advantage of implicit feedback is simplicity in the routers. Routers are left to focus only on resource allocation, and do not have to calculate and produce an appropriate feedback message. However, the scheduling mechanisms must be known to the end hosts for implicit feedback to be useful; otherwise, the observed performance may be misleading and not accurately describe the actual congestion state of the network. For example, with FIFO scheduling an increase in the rate may lead to an increase in the observed throughput, although queues may have already started building up and the total delay has increased.

Explicit feedback can be in the form of congestion notification or rate indication. Due to the limitations in the information that can be carried in protocol headers, explicit feedback can be *binary* or *multi-valued.* In the case of binary feedback, the appropriate operating point is found through an iteration process of network feedback and host adjustments. For explicit feedback, the only methods proposed for TCP/IP networks are Source Quench messages in the Internet Control Message Protocol (ICMP) as well as the Explicit Congestion Notification (ECN) proposal[Ramakrishnan99]. The ICMP Source Quench message is sent by the IP layer of a host or router to throttle back a sender in case the host/router runs out of buffers or throws packets away. ICMP Source Quench is very rarely used in the Internet and the current opinion is to deprecate this message because it consumes bandwidth at times of congestion, and is generally an ineffective and unfair solution to congestion [Floyd94, Stevens94].

In the ECN feedback scheme the router sets a bit in the packet header (Congestion Experience (CE) bit) whenever it detects incipient congestion. The receiver copies this bit into the header of the acknowledgment packet, and the flow control mechanism at the sender is responsible for adjusting the window (or rate) based on a certain algorithm. The algorithms used for congestion detection and window adjustment as responses to explicit feedback are part of the *queue management* and *flow control mechanisms*, respectively. Explicit feedback implies an extra mechanism in the router, but on the other hand provides more quantitative control information, which can be valuable for the adjustment process. *Explicit rate indication* [Charny95] is another method of explicit feedback in which the switches perform rate allocation and the calculated rates are explicitly communicated back to the sources (via the receiver) as information in the packet headers. It has been used in ATM networks but not in the Internet.

1.5.7 Scheduling Mechanisms

Scheduling determines the service order of the packets. Therefore, it is the most direct control over how a network serves its users, and thereby a very effective means of resource management. The scheduling discipline controls the bandwidth allocation by serving a certain number of packets from each flow in a given time interval. Individual scheduling mechanisms will be discussed in detail in Chapter 6. In this part we will give a general overview.

There are several factors that have to be considered in the design of a scheduling discipline. The most important design factor is the *number of priority levels* that may be realized as separate queues. Assuming n priority levels and the higher-numbered levels corresponding to higher-priority users, the scheduler only serves a packet from priority level k if there are no packets waiting to be served at higher priority levels $k, k+1..., n$. The service order within a priority level must also be defined, especially when the priority level serves aggregates of flows with different delay requirements [Keshav97]. This can be something as simple as first-come first-served (FCFS) or more complex by sorting packets based on service tags calculated by the scheduler.

Another design decision is the *degree of flow aggregation* within a priority level. Each flow may be treated separately (per-flow scheduling), or several flows may be treated at the same priority level (class). However, the issue of resource management for flows that belong to the same class still remains, and it is not guaranteed that all flows in the same class will receive the same QoS. The service order within the class and the cooperation between the flows are important here.

Last but not least is the issue of *work-conserving* versus *non-work-conserving scheduling disciplines*. A work-conserving scheduler is idle only when there are no packets that require service. In contrast, a non-work-conserving scheduler may be idle even if there are packets awaiting service. The reason for doing so is that it serves packets only when these become eligible. Eligibility times can be computed in such a fashion that the packet stream after the scheduler conforms to criteria related to packet inter-arrival time variation (jitter) and burst size, which is crucial to the buffer provisioning of the downstream routers. According to [Keshav97], a scheduling discipline must:

- Be easy to implement,
- Provide fairness and protection,
- Satisfy performance bounds(deterministic or statistical), and
- Have easy and efficient admission control procedures.

Sometimes these requirements can be contradictory and will lead to trade-offs. The simplest scheduling algorithm is FCFS, implemented with a first-in first-out (FIFO) queue, which serves packets in order of arrival. With FCFS all users experience the same average delay, even if only a small number of them are responsible for the overload (in other words, greed is rewarded). Thus, FCFS does not protect users and can be unfair too, since it distributes link bandwidth according to queue occupancy distribution. With FCFS, the way buffer space is managed has a direct impact on the way bandwidth is managed. FCFS cannot provide max-min fair allocation. This can be achieved by an ideal work-conserving discipline called *Generalized Processor Sharing* (GPS) [Parekh92b].

GPS serves packets as if they were in separate logical queues, by visiting each nonempty queue in turn and serving an infinitesimally small amount of data from each queue. In any finite time interval, it can visit every logical queue at least once, skipping potentially empty queues. The queues can have weights associated with them and receive service in proportion to their weight, in which case GPS achieves max-min weighted fair sharing. GPS is only a model and cannot be implemented in practice.

The simplest emulation of GPS is *round-robin*, which serves a packet from each nonempty queue instead of an "infinitesimal amount of data". When the queues have weights associated with them they get served in proportion to their weights; the scheme is then called *Weighted Round-Robin* (WRR). In order to allocate bandwidth fairly, WRR requires knowledge of the average packet size for each queue, but this is not realistic given the characteristics of the sources. Moreover, WRR is fair only when examined in time-scales larger than the round-time (the time taken to serve each queue once).

Weighted Fair Queuing (WFQ), [Demers89] is an approximation of GPS that does not require knowledge of the average packet size for each queue. Instead, it emulates GPS by associating each packet with a finish number which corresponds to the time this packet would complete service had it been served by GPS. The packets are then served in order of these finish times and WFQ can provide a flow with QoS guarantees. For token bucket constrained sources and arbitrary topology networks of GPS servers, Parekh and Gallager [Parekh94] proved that there is a bound on the worst case end-to-end delay experienced by a flow that passes through a series of GPS schedulers. The bounds for GPS networks apply to networks of WFQ schedulers for small packet sizes.

The original WFQ is computationally expensive to implement, so several variations of WFQ have been proposed that are optimized for software and/or hardware implementation such as *Worst-case WFQ* (W^2FQ) [Bennett96], *Self-Clocked Fair Queuing* (SCFQ) [Golestani94], and *Deficit Round-Robin* (DRR)

[Shreedhar95].

Scheduling disciplines are a one of the most important part of higher-level resource management mechanisms, like link sharing [Floyd95], which are expected to play a key role as building blocks of future IP service models. Although, strictly speaking, fair queuing may not be absolutely necessary for implementing a certain QoS policy (bandwidth allocation, delay, or jitter guarantee), it is clear that some form of active resource management is required that can provide protection and enforce bandwidth allocation policies.

1.5.8 Buffer and Queue Management Mechanisms

Scheduling by itself cannot offer a solution to resource management inside the network, mainly because traffic can arrive in bursts. Thus, unless there is enough buffer space to absorb these packet bursts and transmit them in subsequent empty intervals, the loss rate can be very high irrespective of the scheduler and by this buffering is essential. However, buffer space in the router is finite and can be exhausted when the traffic arrival rate exceeds link bandwidth for a sufficiently long time. Even if the buffer space were infinite, it would not solve the problem. Nagle [Nagle87] observes that a packet-switched network with infinite buffer, FCFS scheduling, and finite packet lifetimes (based on the *Time-to-Live*, TTL, field in the IP header) in overload conditions will drop all packets.

The role of buffer management is to determine how the buffer space is shared between the different flows that traverse the router and, in particular, those flows that use the same output interface. There is a wide variety of possible strategies for buffer allocation, static or dynamic and based on different criteria like number of flows, current or past bandwidth allocation, and buffer occupancy. The two most popular buffer management schemes are *shared buffer pool* and *per-flow allocation* [Keshav97].

In a shared buffer pool, buffers are used on a first-come first-use basis. There is clearly no protection between the flows since one flow can occupy all the buffers and starve all other flows by simply sending fast enough. Due to its simplicity and implementation efficiency, this scheme is found in most Internet routers today.

Per-flow allocation protects flows from each other by keeping track of buffer utilization and dropping packets based on the buffer occupancy level of each flow. This is considered expensive and cannot scale in terms of processing power to meet the requirements of large numbers of flows in backbone routers. The larger the maximum allowed queue length, the larger the burst size that can be absorbed without dropping packets. However, it is obvious that long queues in the routers increase end-to-end delay.

The role of *queue management* is to control the length of the queue and potentially which flows occupy it, by selecting which packets to drop and determining when this is appropriate. Queue management mechanisms are orthogonal and complementary to both scheduling and buffer management.

Traditionally Internet routers have managed the queues at their links by setting a maximum length for each queue, accepting packets until the maximum length is reached and then dropping subsequent incoming packets until space becomes available in the queue. This method is known as tail drop and has two serious disadvantages – it sustains full queues, and it can cause lockout due to traffic phase effects. It has been realized that lockout phenomena can be avoided by introducing randomization in the network. One such discipline is the *random drop* where a router randomly selects a packet to drop from the queue when a new packet arrives at a full queue. The intention of this discipline is to notify those users whose traffic contributes more to the congestion of the router.

Random drop gateways are reported to achieve improved fairness for late-starting connections and slightly improved throughput for connections with longer *Round-Trip Times* (RTT) [Hashem89]. Random drop is also proposed for congestion avoidance by initiating the dropping of packets when congestion is anticipated instead of only when the queue becomes full. This enhanced version is called *early random drop.*

Early random drop routers are certainly an improvement over drop tail because they alleviate to a great extent the problem of flow segregation. However, they are not sufficient for providing fair bandwidth allocation and cannot successfully contain aggressive sources.

Given the shortcomings of early random drop routers, a new mechanism was proposed by Floyd and Jacobson called *Random Early Detection* (RED) [Floyd93]. RED routers attempt to mark packets sufficiently frequently to control the average queue size and avoid the biases described above. RED is the most prominent and widely studied active queue management mechanism and successfully addresses the problems found in its predecessors. However, it is very difficult to parameterize RED in order to perform well under different traffic conditions. In almost all studies, the parameter settings are based on heuristics, and the proposed configuration is suitable only for the particular traffic conditions studied. It is possible that the performance of a RED router to approach that of a drop tail router for a given set of configuration parameters and traffic conditions.

1.6 Organization of this Work

This work is organized into eight chapters. Beside this introduction part (Chapter 1), an overview of current positioning methods for mobile communication networks is given in Chapter 2. Chapter 3 presents the standardization effort of the main contributors to LBS standardization such as 3GPP, the WAP Forum, the Open GIS Consortium, and OMA (Open Mobile Alliance). Chapter 4 gives a comprehensive overview on the state-of-the-art QoS standardization, mechanisms, and architectures in the IETF and 3GPP, followed by Chapter 5, that proposed a QoS architecture for Location-Based Services in UMTS. Chapter 6 extensively treats different scheduling algorithms, outlining their benefits and limitations. And finally, performance evaluations are presented in Chapter 7, comparing the widely used overprovisioning concept with network-based QoS mechanisms. Chapter 8 conclude this work.

2 Positioning Methods in Cellular Networks

2.1 Introduction

Locating the mobile station is the prerequisite for offering Location-Based Services. Since cellular systems were not originally designed for positioning, the implementation of different location methods may require new equipment to make the necessary measurements for location determination and new signalling to transfer the measurement results to the location determination unit. In this chapter, cellular location methods and their implementation aspects will be presented. There is usually a classification of these methods in network-based, mobile-based, and mobile-assisted methods. They are based on the role of the mobile station and the network or on the location measurement principle [Laitinen01a]. In this work, we make a somewhat different classification, by defining:

- Cell-ID and NMR based,
- Triangulation-based, and
- Satellite-supported methods.

The first group of methods is based on the already existing cell identification information and supplementary information such as propagation time and signal strength measurements. This supplementary information is usually called *Network Measurement Results (NMR)* and together with the *Cell Identification (Cell-ID)* method, it is generally used as the starting positioning method in second-generation cellular networks.

Triangulation-based methods are based on measurements of signal propagation time or the signal angle of arrival between the mobile station and several base stations. For the first group of methods, which are based on measuring the signal propagation time, not less than three different base stations have to be involved for undoubted positioning of the mobile station. Methods in this group are:

- *Time-of-Arrival (TOA)*,
- *Enhanced Observed Time Difference (E-OTD)*, and
- *Observed Time-Difference-of-Arrival (OTDOA)*.

For the Angle of Arrival (AOA) method at least two base stations are necessary. By this, "intersection-based" may be the better notation for AOA than "triangulation-based" method but as we will see, in practice not less than three measurements from three different base stations are used. Otherwise, the positioning accuracy would be too inexact. The positioning accuracy for all these methods will certainly be higher if more than this minimal number of measurements would exist.

The satellite-supported methods use the *Global Positioning System (GPS)* or other satellite systems for positioning of the mobile station. The method described in this work implements additional hardware in the cellular network for assisting the GPS receiver, which has to be integrated into the mobile station. Because of this, the method is called *Assisted Global Positioning System (A-GPS)*.

The previously mentioned network-based and mobile-based classification aims at the question where these measurements are carried out[1]. If they are done in the network, then we are talking about a *network-based* variant of the method. If they are done in the mobile station, it is a *mobile-based* variant. Another further variant originates if the mobile station makes the measurements, but the position calculation is carried out in the network. In this case, we classify this method as a *mobile-assisted* one.

Network-based implementations do not require any changes to existing handsets, which is a significant advantage compared to mobile-based or most mobile-assisted solutions. However, the mobile station must be in active mode to enable location measurements and thus positioning in idle mode is impossible. The AOA method is considered to be network-based only, because it needs an antenna array at the receiver, which presently can only be implemented at the base station side. TOA can be network-based or mobile-based. But in the lit-

[1]In 3GPP specifications they are referred as Position Calculation Function (PCF).

erature, it is usually considered as a network-based method and thereby called *Uplink Time-of-Arrival (UL-TOA)*. In the mobile-based implementation, the mobile station makes measurements and position determination. This allows positioning in idle mode by measuring control channels, which are transmitted continuously. Additionally, this method gives a good solution to the privacy problem concerning LBS, because by this method the subscriber is in charge of the position estimation and the subscriber is in sole possession of the positioning data.

Some assisting information (e.g., base station coordinates) might be needed from the network to enable location determination in the mobile station and the mobile-based implementation does not support legacy handsets. In mobile-assisted methods, the computational burden is transferred to a location center where powerful processors are available. However, signalling delay and signalling load increase compared to a mobile-based solution, especially if the location result is needed at the mobile station. Although mobile-assisted solutions typically do not support legacy handsets, it is possible to use the Network Measurement Results (NMR) that are continuously sent by GSM handsets to the network in active mode. Techniques that use these measurement results (e.g., signal strength and propagation time measurements) are often classified as network-based since they do not require any changes to existing handsets.

The requirements set by different applications may favor different kinds of implementations. For example, emergency call location requires high reliability and it is highly desirable to locate these calls from legacy phones as well as new phones. Applications that use continuous tracking (e.g., route directions) require high accuracy and fast location with a fixed update rate. Since the location result is needed at the mobile station in this case, these requirements are best met with a mobile-based solution. Some applications (e.g., traffic monitoring and Location-Aided Network Planning, LAP), require mass location capability at the network side. These requirements can only be met by network-based or mobile-assisted implementations. Another classification is based on the measurement principle [Syrjärinne01]. The measurement principle of each method belongs to one of three categories:

- Multilateral,
- Unilateral, and
- Bilateral.

In multilateral techniques, several base stations (BTS, Base Transceiver Station) make simultaneous (or almost simultaneous) measurements. The multilateral measurement principle leads to a network-based implementation. Unilateral means that the mobile station measures the signals sent by several base stations and thus leads to a mobile-based or mobile-assisted implementation. For bilateral techniques, multiple measurements are not needed: either the mobile station measures the signal from a single base station or one base station measures the signal from the mobile station. This does not exclude any of the three implementation categories. Since multilateral techniques require coordination of simultaneous measurements at multiple sites, unilateral techniques are generally better in terms of capacity and signalling load. Bilateral techniques are optimal for rural coverage since only one base station is involved.

The combination of all types of solutions (hybrid solution, see Section 2.5) is also used as it enables to combine the advantages of the different techniques while limiting their drawbacks.

2.2 Cell-ID and NMR-based Methods

In the following, the standard Cell-ID method and three proprietary (non-standardized) Cell-ID methods are described. The proprietary methods retrieve the Cell-ID information in different ways. Firstly by using STK (SIM ToolKit) in the handset and sending the information via SMS. Secondly, by reading the information via an IN system. Thirdly, by using the information transmitted with an ongoing WAP call. For all these cell identification methods, Network Measurement Results (NMR) like the propagation time (in GSM known as *Timing Advance, TA*) and signal level measurement (RXLEV measurements in GSM), can be acquired to improve the positioning accuracy. The propagation time measurement is a method to assist all positioning methods and can be used as fall-back procedure[2]. This method has been standardized by ETSI and 3GPP [3GPP04b].

[2] "Fall-back" means that this method will only be used if the primary positioning method cannot be applied.

2.2.1 Standard Cell-ID Method

The simplest method for locating a mobile phone is based on cell identification. Since this is an inherent feature of all cellular systems, minimal changes to existing systems are needed. The identification of the cell only has to be associated with location, i.e., the coordinates of the BTSs must be known (see Figure 2.1a). This is a bilateral location principle that can be implemented as a network-based or mobile-based technique. In the mobile-based implementation, the network would have to continuously transmit the coordinates on a control channel.

Another advantage of this method is that no calculations are needed to obtain location information. The drawback is that accuracy directly depends on the cell radius, which can be very large especially in rural areas (up to 35 km for GSM networks). In dense urban areas, location accuracy is considerably better due to the small cell radius of micro- and picocells. Accuracy can be improved using information of cell coverage area (e.g., sector cells), Timing Advance (TA), and supplementary data (Figures 2.1b, 2.1c and 2.1d).

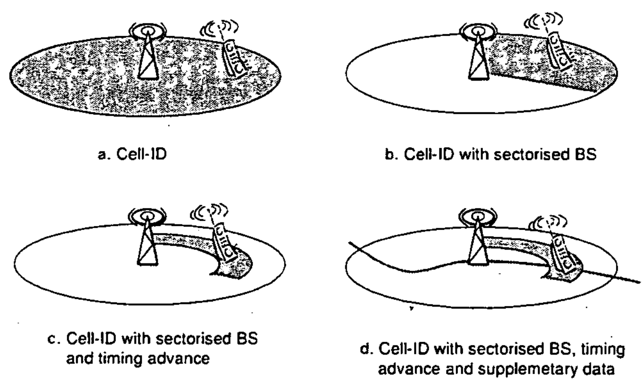

Figure 2.1: Cell-ID method and its extensions

In the GSM case, Timing Advance with the Cell-ID is standardized as a fallback positioning method. In the UMTS interface case, the Cell-ID based method is standardized. The Serving-RNC (Radio Network Controller) determines the identification of the cell providing coverage for the target user equipment (UE). Depending on the operational status of the UE, additional operations are needed:

- Case 1 – User equipment Cell-ID is not known at the time the location request is received at the Serving-RNC: UE should be paged to locate its current Cell-ID. Alternatively, the Cell-ID may be determined as the one that was used during the last active connection to the UE. This determination is accompanied by the time-of-day of the last connection in the cell.
- Case 2 – User equipment Cell-ID is known: If UE is not in soft handover, then the Cell-ID is determined. Otherwise, the Serving-RNC combines the information about all cells associated with the UE. In soft handover, the UE can have several signal branches connected to different cells, reporting different Cell-IDs. A reference Cell-ID is determined by the Serving-RNC based on the coverage area of each cell.

According to [3GPP04c] this Cell-ID should be mapped to geographical coordinates or a Service Area Identifier (SAI), which can include one or several cells. Mapping is accomplished either in the Serving-RNC, in a Network Management System, or by co-operation of various access network elements.

2.2.2 Cell-ID STK Method

The Cell-ID STK (SIM ToolKit) solution uses an STK-script on the SIM card of the mobile station to extract the position information from the network. The serving cell is used to determine the Cell-ID. The position information, Cell-ID and Local Area Code (LAC), is sent to the location manager via the SMS server. The location manager translates the Cell-ID and LAC to a geographical coordinate system by using a database. In addition, the location manager is the interface to the applications and services provided by the operator or any third party. The database of the location manager contains the Cell-ID, LAC, the geographical coordinates and the size of the cell and has to be updated regularly due to changes in the operator's network. The geographical coordinates may correspond to the position of the base station, a point of interest, or the center of the cell. The main disadvantage of the Cell-ID STK solution is the same as for the standard Cell-ID method: the variation of the position accuracy ranging from below 100 m in micro-cells and up to 35 km in rural areas. The analysis of network measurement results, such as signal attenuation and time advance can improve the accuracy.

The method of positioning via Cell-ID STK is rather easy to implement in an existing GSM network without major costs and system upgrades. The needs for this system are a new SIM-card with a location STK script and a location manager for translating the Cell-ID to geographical coordinates using a database. A drawback of the method is the limited and varying position accuracy depending on the size of the serving cell.

2.2.3 Cell-ID IN Method

In GSM, radio contact of the mobile station is in many cases connected to a location update in the Visitor Location Register (VLR). The VLR is updated with the location information of the mobile station. This process assures that the user can be found as soon as a call is requested to the user. The location information, stored in the VLR, contains MCC + MNC + LAC + CI (Mobile Network Code, Mobile Country Code, Location Area Code, and Cell ID). In order to retrieve the location information from the VLR, the VLR can be accessed by the MAP command "Any-Time-Interrogation" via an IN service. The Mobile Application Protocol (MAP) is a standardized protocol in the GSM network. The MAP command requests the location information from the Home Location Register (HLR). The HLR retrieves the data from the related VLR and returns the location information to the IN system.

Contrary to the Location Area Code information, which is mandatory in the VLR, the Cell-ID is only obligatory.

Location determination via IN is a solution, which can be applied to every user independent of the used mobile phone or contractual relationship (postpaid or prepaid). The position accuracy is limited to the Cell-ID. The disadvantage lies greatly in the time accuracy, which limits this position technology only to services for which time and position accuracy can be very poor.

2.2.4 Cell-ID WAP Method

Almost every GSM Network operator in Europe offers a Wireless Application Protocol (WAP) access. The WAP system is directly connected to the switching subsystem of the GSM Network and consists mainly of the WAP Gateway and some security/authentication mechanism including an authentication database. The authentication database contains user data like MSISDN (Mobile Station ISDN Number), IP-Address, and Time Stamp (TS). By adding a data field for the Cell-ID, the location information can be stored in the WAP database. During WAP call set-up, the Cell-ID will be retrieved from the switching network, forwarded to the WAP system, and finally stored into the WAP database. The WAP database can be accessed by the Location Manager.

The access to the location information via WAP is a solution, which can be deployed fast, due to the minor modifications needed in the GSM Network. For services like location information services where the access is accomplished via WAP, this positioning method is optimal. For other kinds of services where the location shall be determined outside a WAP call, the positioning via WAP is not applicable.

Advantages and Disadvantages

The advantages of Cell-ID based methods are that they are easy to implement, they are low-cost, the location fix is fast, the coverage and reliability is very good and last but not least - they support legacy phones.

The disadvantage of Cell-ID methods lies in the insufficient accuracy for many services.

2.2.5 Signal Strength

Signal strength measurements in GSM networks are inherently done together with time propagation measurements. In the GSM system, each mobile station measures the signal levels from up to seven BTSs at 0.48 second intervals to facilitate handover (RXLEV measurements). With software modifications, this information can be used for location purposes but, as we will see, the positioning accuracy by using these measurements is not very high. In addition, a location server is needed to store pre-calculated coverage information and to do location search.

It was one of the first ideas for mobile location to apply signal attenuation measurements [Figel69]. Assuming two-dimensional geometry, an omnidirectional base station antenna and under free-space propagation conditions, signal level contours around a base station are circles. If signal levels from three different BTSs are known, the location of the mobile station can theoretically be determined as the unique intersection point of the three circles (Figure 2.2).

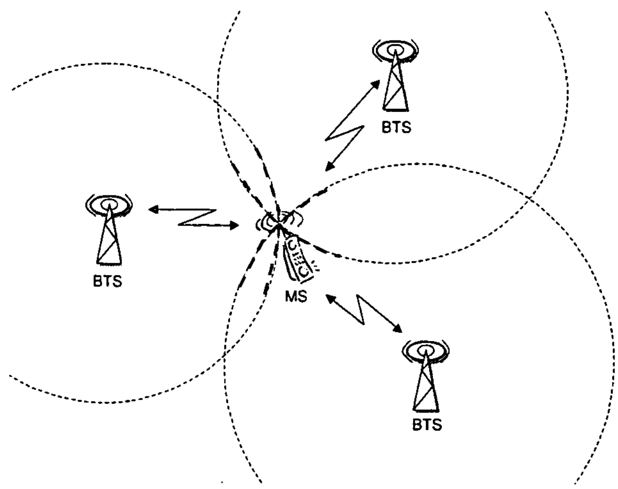

Figure 2.2: Signal strenght measurements from three base stations

However, practical propagation conditions (especially in urban areas) are far from free-space propagation and powerful field prediction tools must be used to compute the signal level as a function of location [Laitinen01a]. Although signal level contours are no longer circles, they can still be used for location estimation by finding the location that produces the best fit between predicted and measured values. An alternative to using a prediction tool is to carry out measurement to obtain a-priori information of the signal level contours. However, this is only feasible in restricted environments, for example as a complement to prediction.

The biggest problem in using signal strength measurements for location lies in the fact that multi-path propagation causes the instantaneous narrowband signal strength to vary by as much as 30-40 dB over a distance of only a fraction of the radio frequency wavelength. Random variations of this order of magnitude cause very large errors in distance estimates. Due to this fast fading, a local average signal level rather than an instantaneous value must be used in location determination. However, mobile terminal averages can be calculated only if the mobile terminal is in motion. Secondly, shadowing by objects near the mobile station causes random variations to the signal strength.

Due to the random variations, signal level based location methods are generally considered unreliable and the expected accuracy of such a technique is poor. However, the accuracy can be enhanced by using a sequence of measurements and smoothing for example by Kalman filtering, as proposed in [Hellebrandt99] where an average location error of 70 m was achieved in simulations.

The signal strength method is unilateral and can be implemented as mobile-assisted or mobile-based method. Mobile-based implementation requires that base station coordinates are transmitted to the mobile station. Signal strength method is easy to implement in GSM, based on measurement reports that are continuously transmitted from the mobile station back to the network in active mode. Therefore, it does not require any changes to existing phones, and is called a network-based method although it is the mobile station that performs the measurements.

An alternative implementation is to modify the mobile station to enable sending measurement reports in idle mode, too. As we have seen, the signal strength measurements are not very accurate but it is an easy and low-cost method to enhance the accuracy of pure cell-ID based location.

In the UMTS downlink, the BTSs send the so-called Common Pilot Channel (CPICH) with constant power of 33 dBm (10% of the maximal power). This pilot channel is unique in each cell and always present in the air. Before any other transmission takes place, each mobile station monitors the pilot channel. Thus, each mobile station is able to measure the power levels of the nearest base station common pilot channels. In UMTS, signal strength measurements may be slightly more reliable due to the wider bandwidth, which allows better smoothing of fast fading. On the other hand, the hearability problem (see Section 2.3.5) prevents measurements of as many neighboring BTSs as it is possible in GSM.

2.2.6 Propagation Time

Propagation time delay measurements between the mobile station and several base stations define the second Network Measurement Results (NMR), method which can be used for location. Assuming two-dimensional geometry and line-of-sight propagation, each time delay measurement defines a circle around a base station and three such circles are needed for unique location determination in the same way as with signal level measurements (Figure 2.2). A problem of this approach is the accurate time synchronization that is required between the mobile station (MS) and the base stations to obtain useful time delay estimates. In GSM, Timing Advance (TA) is a parameter that each base station sends to each mobile station connected to it to ensure that the transmissions of the mobile stations arrive in the correct TDMA time slots. By this the propagation time of the signal between the mobile station and the base station is inherently one of the GSM parameters.

For distances between mobile station and base station under 550 m, the bursts sent by the mobile station arrives exactly into the dedicated time slot without using Timing Advance (Figure 2.3).

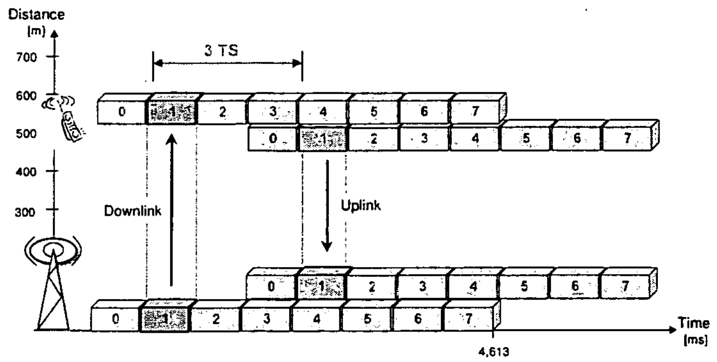

Figure 2.3: Transmission within 550 m range

For larger distances, this would not be the case. As shown in Figure 2.4 illustrating the case without Timing Advance a burst dedicated to time slot 1 and sent by the mobile station, would not be received at the base station into the proper time slot 1, and therefore would interfere with the neighboring time slot 2.

Sending the burst for a time period T_2 earlier, the propagation time between mobile station and base station $(T_1 + T_2)$ enables an undisturbed transmission (Figure 2.5).

The Timing Advance is a 6-bit information for the maximum defined range for GSM which is about 35 km corresponding to about 550 m for the least significant bit of the Timing Advance information. The Timing Advance method is only useful if the size of the cell is bigger than 550 m which is the case

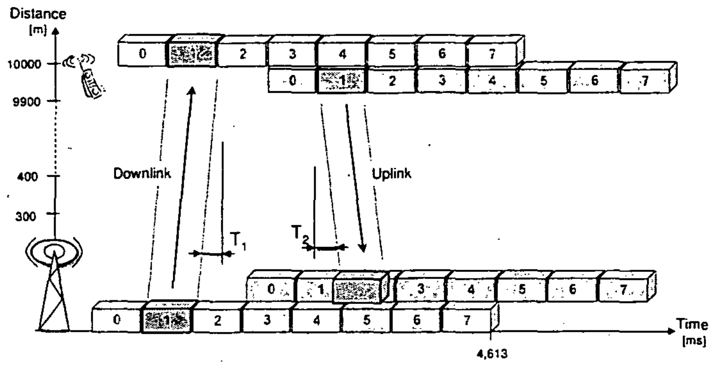

Figure 2.4: Transmission beyond 550 m range without timing advance

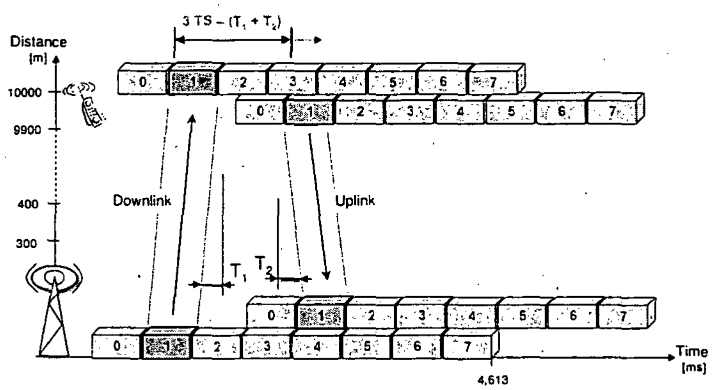

Figure 2.5: Transmission beyond 550 m range with timing advance

in rural and sub-urban areas. Additionally, in the absence of a line-of-sight (LOS) signal component, the measurements are based on reflected signals that may not be coming from the direction of the mobile station. Even if a line-of-sight component is present, multi-path propagation can still interfere with the measurement. Indeed, these are the problems of all time propagation based methods.

The standards [3GPP04b] specify that the Timing Advance positioning method returns the Cell-ID of the serving cell and Timing Advance to the Serving-MLC, where the location estimate is done. The Timing Advance can be used to assist all positioning mechanisms and as a fall-back procedure.

2.3 Triangulation-based Methods

Triangulation-based methods are based on measurements of signal propagation time or the signal angle of arrival between the mobile station and several base stations. For the first group of methods, which are based on measuring the signal propagation time, not less than three different base stations have to be involved for undoubted positioning of the mobile station. Methods in this group are Time-of-Arrival (TOA), Enhanced Observed Time Difference (E-OTD), and Observed Time Difference of Arrival (OTDOA).

Another method would be to measure the angle of arrival of the signal. This method is called the Angle of Arrival (AOA) method. Compared to the first group of methods, two base stations are sufficient for calculating the position of the mobile station. However, the minimal number in practice is three. The positioning accuracy will certainly be higher if more than this minimal number of MS-BTS measurements exist.

The basis for E-OTD and OTDOA is the estimation of the time difference of signal arrival, described in Section 2.3.3.

2.3.1 Time of Arrival (TOA)

The Time of Arrival (TOA) method is based on signal measurements performed either at the base stations or at the mobile station. If the base stations and the mobile stations are synchronized, TOA measurements are directly related to the distances between base and mobile station and measurements with three different base stations are needed for the unique positioning in two dimensions[3] (Figure 2.6).

However, if the network is not synchronized, as is the case in GSM and UMTS on the UTRAN-FDD interface, TOA measurements can only be used together with a common time reference for the base stations. In this case, the TOA measurements are not directly related to the MS-BTS distances but two TOA measurements then define a hyperbola, and four measurements are needed for unambiguous 2D location (Figure 2.7).

The common time reference can be conducted by using a GPS receiver at

[3] To determine the altitude a fourth measurement point is needed.

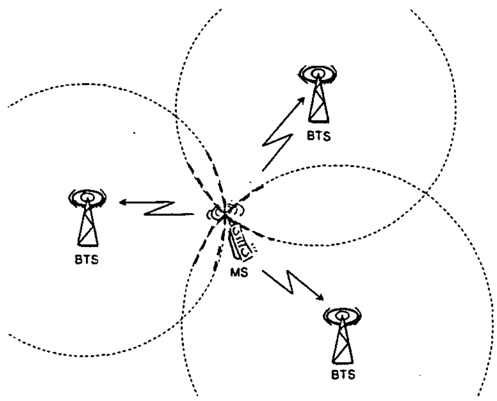

Figure 2.6: Time of arrival measurement method with synchronized base stations

each base station site. If measurements are performed at the base station, it is a network-based multilateral technique and called Uplink Time-of-Arrival (UL-TOA). Thereby the part in the base station in charge of making the measurements is called *Location Measurement Unit (LMU).* It can be placed in the network together with the base station or stand-alone. It was taken into GSM standardization as a candidate E911 solution. In the GSM implementation of UL-TOA, the location of an mobile station with an on-going call is accomplished by forcing the mobile station to request a handover to several neighboring base stations. The signal sent by the mobile station consists of up to seven access bursts at full power.

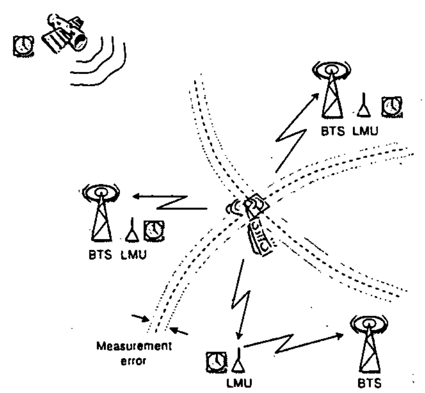

Figure 2.7: Time of arrival measurement method

It is intended from the beginning that the handover (intra- or inter-cell) will fail. The time of arrival of the access burst signal is measured in the LMUs associated to the base stations. From the Cell-IDs, time-of-arrival values, and the time-of-arrival measurement quality information returned to the Serving-MLC, the position can be calculated via hyperbolic triangulation by using a *Position Calculation Function (PCF)* (see Annex A). The position accuracy is limited due to the multi-path propagation and the LMU measurement errors.

Network-based solutions of TOA have two drawbacks compared to downlink methods: it is only possible to perform the measurements in dedicated mode and there may be capacity problems due to the multilateral measurement principle. The advantage is that due to the network-based implementation, UL-TOA supports legacy phones, though a channel may need to be reserved for the attempted handover.

Advantages and Disadvantages

The advantage of the TOA method is that it supports existing GSM phones.

The disadvantage is that mobile and base station have to by synchronized or Location Measurement Units have to be integrated into the network. Although synchronization of the base stations is achievable, the real problem is the synchronization of the mobile station. The number of LMUs is about the same as the number of base stations and thereby additional costs should not be underestimated. Additionally forced handovers are needed to make TOA measurements.

The next disadvantage is common to all triangulation methods. If the base stations are in a line, triangulation is much more difficult and therefore may be imprecise. This is often the case in rural areas along highways.

2.3.2 Angle of Arrival (AOA)

Angle of Arrival (AOA) is based on triangulation, where the angle of arrival is determined by electronically steering the main lobe of an adaptive phased-array antenna in the direction of the arriving signal. Usually, by using a complex array of antenna elements at the base station side the direction of the arriving handset signals can be measured at the base station and thereby the AOA method is classified as a network-based method. When angles of arrival are computed for several cell sites, the location of the mobile stations can be estimated based on the point of intersection of projected lines drawn out from the cell sites at the angle from which the signal originated. Three or more angle estimates are needed to resolve any ambiguities (Figure 2.9). This is because by combining only two AOA estimates there would be a large uncertainty associated with any measurement of a mobile being close or on the line that connects the base stations performing the measurements (Figure 2.8).

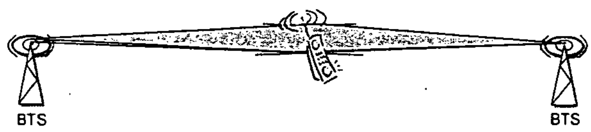

Figure 2.8: Angle of arrival method with two base stations

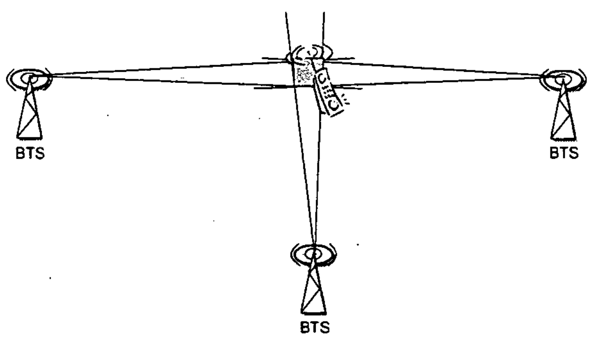

Figure 2.9: Angle of arrival method with three base stations

This method is well suited for tracking a continuous signal but it is fairly difficult to determine the direction of short digital signals [Baumann01].

The accuracy of the measurement is severely degraded if there is no clear line-of-sight (LOS) between the antenna array and the handset (multi-path reflections) and it depends on the number of available measurements and the geometry of base stations around the mobile station. In addition, the accuracy is reduced as the mobile unit moves away from a cell site, since the arc length becomes greater even though the angle of arrival remains constant. However, the AOA technique could be used in rural and suburban areas where the attainable accuracy is better and it is an advantage to be able to locate a mobile station that can only be measured by two BTSs.

The AOA method for positioning a mobile subscriber unit is currently not standardized in the GSM system. The main disadvantage of the AOA method is the requirement for complex antenna arrays consisting of 4 to 12 antennas at a spacing of less than one radio frequency wavelength (900 MHz: 33 cm; 1800 MHz: 16 cm). In addition, a precise calibration and calibration maintenance is needed. The estimated position accuracy is above 125 m and the time needed for positioning is about 10 seconds [Baumann01].

Another possibility of implementing AOA would be the use of smart antennas. Embedded smart antennas use adaptive array processing within the channel elements of a base station. The smart antenna processing takes place in the signal path of the base station, using a customized, narrow beam to track each mobile in the network [Lehne99].

Advantages and Disadvantages

The advantages of the AOA methods are that two-dimensional location is theoretically possible with only two AOA measurements; they are relatively accurate in good conditions (mobile station near to the base station) and they support legacy handsets.

The disadvantages are that they need complex and expensive antenna arrays at the base stations. The complexity of the AOA algorithms is because of the need for measurement, storage, and usage of array calibration data. Reduction in accuracy occurs with increasing distance. Furthermore there are also potential capacity problems.

2.3.3 Time Difference of Arrival (TDOA)

Time Difference of Arrival (TDOA) is a method which determines the mobile station position based on trilateration by using time difference measurements rather than absolute time measurements as TOA does. This method is the basis for two positioning methods:

- E-OTD, Enhanced Observed Time Difference – specified as one of the GSM positioning methods.
- OTDOA, Observed Time Difference of Arrival – developed by 3GPP members for the UMTS Terrestrial Radio Access Network (UTRAN) [3GPP04c].

In both methods, TDOA is realized as unilateral mobile-based or mobile-assisted method. The TDOA method calculates the time interval between the receptions of bursts from two different base stations. Assuming that bursts from BTS_1 and BTS_2 are received at time t_1 and t_2, respectively, the *Observed Time Difference (OTD)* can be calculated by

$$OTD = t_2 - t_1 \tag{2.1}$$

In a synchronized network it is enough to measure the OTD value to determine the position, but GSM, UMTS in FDD mode and most other cellular networks are not synchronized[4]. To deal with this, a parameter called *Real Time Difference (RTD)* is defined. The RTD value gives information about the synchronization difference between base station sites, and is defined as

$$RTD = t_4 - t_3, \tag{2.2}$$

where t_3 is the transmission time of BTS_1 and t_4 the transmission time of BTS_2. To measure the RTD values, separate measurement units are needed and placed in a fixed location. These units are called *Location Measurement Units (LMU)* . If they are stand-alone, they are LMUs of type A and if they are integrated with a base station, they are LMUs of type B .

One more parameter is needed - the *Geometrical Time Difference (GTD)* . This parameter describes the effect of the geometry on the burst reception time and is defined as

$$GTD = (d_2 - d_1)/v, \tag{2.3}$$

where d_1 is the length of the propagation path between BTS_1 and mobile station, d_2 the length of the propagation path between BTS_2 and mobile station

[4]UMTS in TDD mode is synchronized.

and v is the speed of radio waves. The relation between OTD, RTD, and GTD is given by

$$OTD = RTD + GTD \tag{2.4}$$

GTD is the actual quantity that is useful for location purpose, since it contains information about the position of the mobile station. If only OTD values are known, no location can be calculated, thus also RTD values must be known. The mobile station location is calculated from the GTD based on the fact that the possible location for the mobile station observing a constant GTD value

$$d_2 - d_1 = constant$$

between two base stations is a hyperbola. Therefore, the TDOA method is often referred to as the *hyperbolic system*. The mobile station can be located in the intersection of two hyperbolas obtained with three base stations and two GTDs. If more GTDs are available the possible location area can be reduced.

The grey area in Figure 2.10 represents the area of uncertainty for the mobile station based on the OTD measurement. The intersection of the hyperbolas is the calculated most likely location for the mobile station.

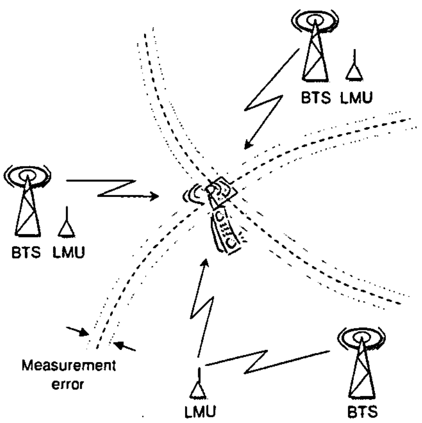

Figure 2.10: Time difference of arrival method

The calculation of the position is done by the *Position Calculation Function (PCF)* (see Annex A), which can be placed in the network or in the mobile station. Once again, if placed in the network, this kind of positioning is called *mobile-assisted positioning*, because the mobile station only delivers the necessary measurement data, but does not calculate the position on its own. If the PCF function is realized in the mobile station it is called *mobile-based positioning*. In the mobile-based mode, the mobile station makes the measurements and also carries out the location calculation, and thus requires additional information (such as the physical location information of the measured base stations) for the location calculation.

The accuracy of the system is also a function of the relative base station geometric locations. The best results are achieved when the base stations is equally surround the mobile station. If they do not, there is a reduction in accuracy, which is termed the *Geometric Dilution of Position (GDP)*. It also requires either precisely synchronized clocks for all transmitters and receivers or a means to measure the time differences, such as the use of LMUs. Otherwise, a 1-μs timing error could lead to a 300 m position error. Due to this, the synchronization must be done to a level of accuracy of the order of tens of nanoseconds.

Advantages and Disadvantages

The advantage of all TDOA-based methods, compared to the TOA method, is that only the base stations have to be synchronized since the time difference in the reception of the same mobile originated signal is measured. Additionally, they require less LMUs and use fever network resources, they are relatively accurate, measurements can be done in idle mode, and if realized mobile-based they provide a good privacy protection.

The disadvantages are the necessity of using LMUs, software modifications to mobile stations, and the hearibility problem in CDMA networks.

2.3.4 Enhanced Observed Time Difference (E-OTD)

Enhanced Observed Time Difference (E-OTD) is a TDOA-based method specified as one of the positioning methods for GSM. The E-OTD approach involves the handset in estimating the timing difference between the various base stations. The handset scans the transmissions of the BCCH (Broadcast Common Control Channel) carrier from a number of base stations (using either normal burst, synchronization burst, or dummy bursts, each with their respective known training sequence) to estimate a time of arrival according to its own internal clock. However, no absolute timing stamp is required like in the TOA method, but the mobile directly estimates the timing difference (OTD). In addition, the handset can carry the measurement process both in idle mode as well as during a call.

As we know from TDOA, the timing difference measured by the mobile station does not represent range differences to surrounding base stations since the time of launch of the signal is not necessarily synchronized. There is the choice of imposing timeslot synchronization at the base-station radio level (i.e., additional GPS receiver at each base station) or alternatively deploying reference units at known locations whose function is to measure the time difference.

To obtain accurate triangulation, OTD measurements for at least three geographically distinct base stations are needed and also, for non-synchronized base stations, RTD measurements are needed. Measuring the OTD values, the location of the mobile station can be calculated either in the network (mobile-assisted positioning) or, if all the needed information is available, in the mobile station itself (mobile-based positioning). A Position Calculation Function (PCF), located either in the mobile station or in the network, determines the location estimate.

Advantages and Disadvantages

E-OTD is one of the most accurate positioning methods on the market today. Thereby it is also specified as a positioning method for GSM, GPRS, and UMTS. It is additionally the first choice of American GSM network operators for the realization of the E911 service. At present, the disadvantage are related with the cost of deployment of the technology, which revealed to be above expectations after the implementation in the US following the E911 regulation entering into force.

2.3.5 Observed Time Difference of Arrival (OTDOA)

The Observed Time Difference of Arrival (OTDOA) method is being developed by 3GPP members for the UMTS Terrestrial Radio Access Network (UTRAN)

[3GPP04c]. Therefore, we use the 3GPP terminology in this section, calling the mobile station as user equipment (UE), the base station as Node-B, and the base station controller as Radio Network Controller (RNC).

OTDOA is a TDOA-based positioning method. This method determines the user equipment position based on triangulation by using time difference measurements. Since these measurements are based on the signals from the Node-Bs, the locations of these Node-Bs are necessary for the network or user equipment to calculate the position. If the Node-Bs are unsynchronized, the Real Time Difference (RTD) between the transmissions of the Node-Bs at the time the OTDOA measurements were made must be provided. In the 3GPP specifications, this is called SFN-SFN (System Frame Number) observed time difference and measured by the UE, which identifies the time difference between two cells as TDOA. One way to obtain these measurements is to deploy Location Measurement Units (LMU), which perform timing measurements of all the local transmitters in fixed locations of the network. These measurements can be converted to real-time difference and then transmitted for positional calculation to the user equipment (UE-based positioning) or Radio Network Controller (UE-assisted positioning).

Two types of these measurements have been defined. Type 1 is used for soft handover and type 2 is used for positioning. The main difference between these two types is that type 2 is applicable for both idle and connected modes, while type 1 supports intra-frequency measurements for the connected mode. Since Node-Bs in the Time Division Duplex (TDD) mode are generally synchronized, the RTDs are typically constant. For the Frequency Division Duplex (FDD) mode, RTT[5] and the Rx-Tx time difference can be obtained to improve performance. Since adaptive antennas have been specified as a feature for 1.28 Mchip/s TDD networks, AOA can be used to further improve the OTDOA performance [Zhao02].

The OTDOA location method in UTRAN has its problems, such as hearability, unsynchronized Node-Bs for FDD mode, geographic location of the Node-Bs, and capacity loss.

[5]Round Trip Time; similar to TA in GSM.

Synchronisation issues

Synchronisation must be done to a level of accuracy of the order of tens of nanoseconds (as 10 nanoseconds uncertainty contributes an error of 3 meters in the location estimate). Drift and jitter in the synchronization timing must also be well controlled as these also contribute to uncertainty in the location estimate.

Generally, the Node-Bs are synchronized in the TDD operating mode. Alternatively (typically in FDD mode), the Node-Bs may be left to free run within some constraint of maximum frequency error. In this scenario, the RTD will change (slowly) with time. The rate of change will depend on the frequency difference and jitter between Node-Bs. If, for example, the maximum frequency difference between two Node-Bs is 10^{-9}, then the start of transmission of a 10 millisecond code sequence will drift through a cycle in about 57 days [3GPP04c]. With this relatively slow rate of drift, the RTD can be measured by fixed units at known locations (LMUs) and stored in the database for use by the Position Calculation Function (PCF). Ongoing measurements of the RTD may be made to assure that the most recent values are available for the PCF.

Hearability Problem

In some conditions a sufficient number of downlink pilot signals may not be available for measurement at the UE. This may occur, for example, if the UE is located quite close to the UTRAN transmitter and its receiver is blocked by strong local transmissions. This is referred to as the *hearability problem*. The hearability problem is a basic consequence of a CDMA radio system. A terminal near its serving Node-B cannot hear other Node-Bs on the same frequency. In order to calculate terminal location, the terminal should be able to receive at least three Node-Bs. Another problem is capacity loss. The signalling related to location calculation may take capacity from other services. This capacity loss should be minimized.

Idle Period Downlink (IPDL)

A solution for the hearability problem is the *Idle Period Downlink (IPDL)* method. With this method, each Node-B ceases its transmission for short periods of time (idle periods) in continuous or burst mode. In continuous mode, the idle periods are active all time and one idle period is placed in every DL frame (10 ms). In burst mode, the idle periods are arranged in bursts and an idle period spacing is under the operator's selection, e.g., one IPDL every 10 frames (100 ms). During these idle periods, the serving base station completely ceases its transmission and the mobile station is scheduled to take the needed OTDOA measurements (SFN-SFN) from the neighbour BTSs, which is now hearable. In addition, real-time difference measurements can be carried out during the idle periods. The idle periods are short and arranged in a pseudo random way. With longer idle periods, the achievable accuracy would be better because of longer integration time at the mobile station, but the system capacity would be reduced.

By supporting the IPDL, OTDOA performance in the mobile station will improve, as there will be less interference during idle periods. Idle periods in the downlink are standardized for the OTDOA-IPDL method. However, the support of the idle periods is optional for the MSs. Because the IPDL method is based on the forward link (downlink), the location service can be provided efficiently to a large number of terminals simultaneously.

Time Aligned Idle Period Downlink (TA-IPDL)

Time Aligned Idle Period Downlink (TA-IPDL) is a specific configuration of standard IPDL where the idle periods of the different sites are intentionally time aligned approximately 30 μs across the Node-Bs. This creates a common idle period during which all UEs are scheduled to make their measurements. During the common idle period, each Node-B will either cease transmission entirely (as in IPDL) or transmit the common pilot. In simulations by [Lopes00, Cruickshank00], the interference level is noticed to be lower for TA-IPDL than for IPDL. Due to lower interference, TDOA estimation is more accurate, more base stations will be hearable to the mobile station and multipath rejection is more effective. TA-IPDL reduces the handset complexity, but additional signalling is needed as well as added complexity in the network. In [Cruickshank00], it has also been noticed that when increasing the number of measured base stations without making line-of-sight state estimations before location estimation, the accuracy is reduced. This is due to the increased probability of using non-line-of-sight measurements, which degrade the location estimation accuracy.

The results show that time alignment can be used to improve positioning performance not only in interference limited conditions, but also inside buildings as well as in the non-homogeneous network layouts which are likely to be typical in early deployments.

2.4 Satellite Supported Methods

The location approach based on a mobile phone integrated with a satellite receiver such as GPS (Global Positioning System) is attractive since GPS achieve good accuracy, at least in open areas. The disadvantage is that buildings attenuate the satellite signals. Therefore, reception of an adequate number of satellites in dense urban areas or within buildings may be difficult and indoor positioning even impossible. Anyway, a considerable effort is being put on combined cellular/GPS technology and integrated GSM/GPS phones have already been introduced. Especially the so-called network-assisted GPS or Assisted-GPS (A-GPS) methods have received considerable attention lately and have been taken to UMTS standardization. In this section, we will present GPS, Differential-GPS, and Assisted-GPS. Also further options like GLONASS, EGNOS, Galileo, and Loran-C will be described shortly.

2.4.1 Global Positioning System (GPS)

GPS is a space-based radio positioning system that provides 24 hours three-dimensional position, velocity, and time information to suitably equipped users. The basic measurement performed by a GPS receiver is the time required for any signal to propagate from one point to another. Because in the general case, the velocity of radio frequency signals is known with relative accuracy, this time measurement can easily be converted to distance-range from the radio frequency source. If the range from the receiver to four satellites is calculated, the receiver can accurately determine his position anywhere on earth. Unlike previous navigation systems using ground-based transmitters, satellite-based transmitters are used to cover earth with higher accuracy than that available from the land-based systems.

The GPS receiver contains a specialized computer that calculates the location based on the satellite signals. The user does not have to transmit anything to the satellite and the satellite does not know that the user exists. There is no limit to the number of users that can be using the system at any one time.

The GPS system consists of at least 24 satellites operated by the United States Air Force, under the control of the Department of Defense (DoD). The first GPS satellite was launched in February 1978. Meanwhile there are five different types of satellites, named Block I, Block II, Block IIA, Block IIR, and Block IIF. Since GPS was designed with main consideration on military applications, the DoD needed to have a way to limit its accuracy in order to prevent the technology from being used in a non-peaceful manner. Therefore, the DoD decided to incorporate errors into the signal received by civilians, thus intentionally degrading its accuracy. The relative process is called Selective Availability (SA).

Several issues affect the effectiveness of location services. These issues are accuracy, response time, time to first fix, service coverage, and integrity. The main advantages of GPS-based systems are the global coverage and the good accuracy (20 - 100 m), especially without Selective Availability (SA) degradation ($<$ 20 m), besides the minimum impact on existing communication networks. On the other side there are problems concerning the weakness of the GPS signal inside buildings, power consumption, and the time to get the first fix at the start up of the receiver. The Department of Defense has commissioned a new set of improved GPS satellites (Block IIF). The deployment of these satellite had begun in 2003, the first lunch is scheduled for 2005, and the new signal will be available to civilian users in 2012, with an accuracy range of 2 to 6 meters [Rao03].

2.4.2 Differential GPS (D-GPS)

The purpose of Differential GPS (D-GPS) is to improve the accuracy of the standard GPS positioning erasing or dramatically reducing some kinds of errors that affect GPS solution like ionospheric errors, tropospheric errors, code measurement errors, and satellite clock inaccurateness. This may be accomplished taking into account that over limited distance between a reference station and remote receiver, the errors will be the same at either location and will consequently cancel out. The method makes use of two GPS receivers; one stationary and located at a known point (reference receiver), the other is operated as a mobile receiver. Since the coordinates of the reference receiver are known, it can correct satellite pseudo-range to the true range. True range less the pseudo-range is equal to the differential correction. Through communication between the reference and mobile receivers, the mobile receiver's pseudo-ranges are corrected to true ranges. There are basically two methods to apply differential corrections and get a better accurate position of mobile receivers:

- Direct Mode – broadcasts the differential correction to the mobile receiver (e.g., via a GSM network) and it is in charge of the application of the correction and the computation of the positioning solution. This method is generally used by government agencies such as US Coast Guard to provide better positioning capabilities to a large and unknown number of boats and ships using different GPS equipment.

- Inverse Differential GPS – consists of the transmission of the "raw" measurements from the mobile receiver to the reference station, where the positioning fix is performed. This kind of solution is very useful when the mobile user has no need to know his exact position and this task is left to the control center. Indeed the control station has generally powerful processing equipment and the bandwidth needed for the data transmission is very low. This method is widely used by service providers offering commercial assistance services to car drivers, exploiting GPS and GSM network capabilities.

The advantage of D-GPS positioning compared to the GPS standard is the higher accuracy that can be reached (1 to 7 meters depending on Selective Availability and baseline between mobile receiver and reference station). However, D-GPS is affected by the usual disadvantages of satellite positioning systems (poorly penetration inside buildings, long time to first fix, etc.).

2.4.3 Assisted GPS (A-GPS)

Assisted GPS (A-GPS) is basically a variation of differential GPS in which the cellular networks (GSM/PCS) are used to deliver the additional signal. It therefore implies a distributed architecture: for instance a GPS reference receiver, a location server, and a GPS enabled mobile station.

Any GPS positioning system has four principal functions:

- Determining the code phases (pseudo-ranges) to the various GPS satellites,
- Determining the time of applicability for the pseudo-ranges,
- Demodulating the satellite navigation message, and
- Computing the position of the receiving antenna by using these pseudo-ranges, timing, and navigation message data.

Most commercial GPS receivers perform all of these operations without any external assistance. In these conventional receivers, the satellite navigation message and its inherent synchronization bits are extracted from the GPS signal after it has been acquired and tracked. Collecting this information normally takes thirty seconds to several minutes, and a high received signal level is required. In an assisted GPS system, these functions are distributed among a GPS reference receiver, a location server, and a GPS-enabled mobile station.

A GPS reference receiver gathers navigation messages and differential correction data for all visible satellites. In another system configuration, the GPS reference receiver may be replaced by a network of reference receivers to provide coverage for a wider area such as the continental USA or Europe. The location server receives and stores data from the GPS reference receiver (or network), provides assisting data to the mobile units, and performs navigation solutions upon receipt of pseudo-range measurements from the mobile station. The assisting data, sent to each mobile station on-demand, consist of a list of satellites in view from the mobile station and their relative Doppler offsets.

This small message (about 50 bytes) is all the mobile station needs to know from the location server to extract pseudo-range information from its short snapshot of GPS data. The server also could have access to a terrain elevation database and this allows it to perform accurate altitude aiding for ground-based applications, a capability that is extremely difficult to implement in mobile-based solutions. The terrain elevation provides practically an extra range measurement, improving reliability and accuracy. Moreover, the server could mitigate multi-path and reflected signal effects using several sequential measurements optimization techniques.

The GPS-enabled mobile station can track far weaker GPS signals than a conventional GPS receiver because it does not need to decode the navigation message. Indeed finding these weaker signals accurately and quickly requires a powerful signal processing element to search over the large number of Pseudo Random Noise (PRN) codes, times of arrival and offset frequencies, and this is done by the server. Finally, we can sum up the advantages of Assisted-GPS positioning systems as compared to GPS standard in higher accuracy, smaller delay for a first fix, reduced power consumption and the capability to operate in environments where the GPS signal would be too attenuated to be useful such as urban areas and inside buildings.

2.4.4 Further Options

GLONASS

The Russian Global Navigation Satellite System (GLONASS) is based on a constellation of active satellites, which continuously transmit coded signals in two frequency bands. The coded signals can be received by users anywhere on the earth's surface to identify their position and velocity in real time based on ranging measurements. The system shares the same principles with GPS concerning data transmission and positioning methods. Nowadays GLONASS constellation consists of 11 active satellites so it is virtually impossible to use this system stand-alone. Nevertheless, receivers able to decode both GPS and GLONASS signals can take advantage on basic GPS receivers positioning by exploiting the increased number of satellites available, particularly in the urban environment.

Though dual receivers are currently available, they do not exhibit wide diffusion and their cost is very high, so this system is not particularly attracting.

EGNOS

The European Geostationary Navigation Overlay Service (EGNOS) is a satellite- and ground-based system that augments the existing satellite services provided by the American GPS and the Russian GLONASS for those users who are equipped with an appropriate receiver. The aim of EGNOS is to augment the GPS and GLONASS signals with additional information (ionospheric correction data), which will ensure the accuracy and integrity for multimodal transport applications. Correction data will improve the accuracy of the current services from about 20 m to better than 5 m. The EGNOS signal is acquired through an infrastructure based on space and ground segments (three transponders installed in geostationary satellites, a ground network of 34 positioning stations, and four control centers). The EGNOS System will provide Europe with three service augmentations to the GPS and GLONASS systems:

- **Geo-stationary ranging service**, which will provide the users with additional pseudo-range measurements,
- **Ground Control Integrity Monitoring** (GCIM) **service**, which will improve the integrity of the Navigation Services to the users, and
- **Wide Area Differential** (WAD) **service**, which will improve the position accuracy by broadcasting correction to users.

This kind of information can also be transmitted on demand to users via the terrestrial communication network (e.g., GSM). The planned date of service availability is the first Quartal of 2005 and it would last up to 2019. The main advantage of this system is the fast integrity information availability which makes it suitable to safety-critical transport and life applications.

Galileo

Galileo is the new satellite navigation system sponsored by the EU and is being undertaken by the European Space Agency (ESA). It is scheduled to come on-line during 2008. It has some significant advantages over GPS in that:

- It has been designed as a non-military system and so can provide a high level of continuity of service, uninterrupted by military requirements.

- The constellation of satellites is designed to provide a potentially higher level of accuracy than GPS and to extend coverage to areas at extreme latitude.

- It sends an integrity signal informing users immediately of any errors that occur.

- It will complement GPS since a receiver will typically work on both GPS and Galileo signals. This will result in enhanced precision and a higher confidence of service continuity.

3GPP is including the Galileo system in its standards as one of the technologies that will be used to determine the geographic position of a user to enable location services in UMTS Release 6. Galileo is also a source of high-precision timing information. One believes that this timing information can be used to enhance the synchronization of the UMTS base-stations and hence increase the capacity of the system.

Loran-C

Loran is an acronym for Long Range Navigation and uses long wave (90-110kHz) transmissions from a chain with three or more terrestrial transmitting stations to compute a position. "C" stand for version C.

Loran-C is installed world-wide and supports a large user community. It provides 0.25 nautical miles (674 m) absolute accuracy, $18 - 90$ m repeatable accuracy (95% confidence), and 99.7% availability. In addition to providing a navigation service, Loran-C transmissions are also used for time synchronization applications (e.g., for communication networks). Furthermore, the Loran-C signal can be modulated to broadcast differential GPS correction data and GPS integrity information. This service is implemented under the name Eurofix [van Willigen98]. Loran-C receivers can operate in two modes:

- Measurements of time differences between different stations (hyperbola mode).

- Measurements of time-of-arrival of the signal (Time of Emission control mode, TOE).

The Loran-C system was developed in the 1950s by the US Department of Defense. After the implementation of GPS, the Loran-C stations outside the USA were handed over to the national governments. The European stations are now operated by NELS (North West European Loran-C System). The member states are: Norway, France, Denmark, the Netherlands, Ireland, and Germany. This development led to a modernization of the European Loran-C system. The experience that GPS could not be used solely for safety critical applications and a complementary system was required, led to enormous upgrade of the US Loran-C system including funding. In other parts of the world, Loran-C stations or similar systems are under operation.

Loran-C offers the advantages that the signal can be received even inside buildings and the Time of Emission control mode allows combined use with GPS in the way that position solutions can be calculated even if one system cannot provide a position solution.

2.5 Hybrid Solutions

Hybrid location techniques combine several of the methods described above to provide positioning estimates with better accuracy, reliability, and coverage. They including indoor, outdoor, urban, and rural areas. The hybrid techniques are not standardized and not all the signalling needed in the network may be available. The drawbacks of hybrid systems are usually greater processing requirements and increased network costs. Usually using a hybrid solution means involving two techniques. Thus, the cost will be as high as using two separate solutions. In the following, some ideas to implement hybrid solutions are presented.

2.5.1 AOA + RTT

A potential UMTS location technique especially in rural and suburban areas where a LOS connection between the UE and the serving Node-B is often active is the Angle of Arrival - Round Trip Time (AOA-RTT) hybrid. By this method, even one base station is enough for location estimation. It is a bilateral network-based method that avoids the hearability problem since a single base station, equipped with an antenna array, can make the necessary measurements. The location estimate accuracy of this technique is limited by the beam width of the antenna array and RTT resolution. As with the AOA method, the location error will increase with distance.

2.5.2 OTDOA + AOA

In UMTS, the OTDOA measurements will be available in every mobile station and deployment of antenna arrays will enable the AOA measurements without extra costs. The performance of both OTDOA and AOA techniques is decreased due to non-line-of-sight (NLOS) conditions. Even though the errors in AOA measurements (originating from non-line-of-sight conditions) are correlated to the errors affecting the timing measurements involving the serving base stations, they should be useful to the location estimation. In [Cruickshank01], the UMTS system using TA-IPDL has been simulated and the results show an improvement of 20% – 60% in location error performance when using the available AOA data in rural, suburban, and urban car scenarios.

Using the OTDOA-AOA hybrid the mobile station positioning may be made possible even in highly non-line-of-sight conditions or by measuring only two base stations. The accuracy of the hybrid is better than OTDOA or AOA alone and the coverage increases if two base stations are enough for location. In addition, it avoids problems with high Geometric Dilution of Position (GDP), e.g., in a highway scenario where the base stations are aligned with the highway. In this case, pure AOA positioning would suffer from dilution of precision.

2.5.3 GPS + E-OTD/OTDOA

This hybrid method in some cases can reach a better level of accuracy and wider coverage and in some other cases provides better availability than either method separately. E-OTD/OTDOA method has better accuracy and availability in urban areas and in-door rather than rural or sub-urban. Meanwhile, GPS has better accuracy and availability in rural and sub-urban areas than urban. However, GPS has no coverage indoors. This hybrid technique is technically possible in GSM standards since E-OTD and GPS reports can be retrieved by successive requests.

2.6 Other Location Methods

2.6.1 Database Correlation Method (DCM)

The Database Correlation Method (DCM) is a generic location method that can be applied to any cellular network [Laitinen01b]. The key idea is to buffer the signal information seen by a mobile station, from the whole coverage area of the location system, in a database that is used by a location server. The database should contain signal information samples, called *fingerprints*, with a resolution comparable to the accuracy that can be achieved with the method, and this resolution may vary in different environments. Depending on the particular cellular system, the signal fingerprints could include signal strength, signal time delay, or even channel impulse response. Any location-dependent signal information that can be measured by the mobile station is useful for the DCM technique. In addition, it is possible to use measurements performed by the network as well as by the mobile station. When the mobile station needs to be located, the necessary measurements are performed and transmitted to the location server. The location server then calculates the mobile station location by comparing the transmitted fingerprint and the fingerprints of the database.

DCM can be implemented in any wireless system, the mobile station only needs to be able to transmit a location-dependent fingerprint to the location server. This fingerprint may consist of signals measured from GSM, UMTS, and/or GPS. The location server must be powerful enough to process all location requests within a reasonable time. In a large-scale implementation, this may require distributed processing.

The major effort in applying DCM is the creation and maintenance of the database. The signal fingerprints for the database can be collected either by measurements or by a computational network planning tool. Measurements are more laborious but produce more accurate fingerprint data. Furthermore, a combination of measured and computed fingerprints can be used. The compensation for the effort to build the database is an optimal location accuracy in environments where the assumption of line-of-sight propagation is not valid, e.g., in dense urban and indoor environments. The only assumption is that the database contains up-to-date data. However, minor changes in the network or propagation environment, e.g., new buildings, will only be seen as lowered location accuracy if the database is not updated. Additionally, it should be noted that similar information that is contained in the DCM database is also needed in network planning. Therefore, the creation and maintenance of the database also supports network planning. The advantages of this method are its good accuracy in urban areas, ability to locate the phone using a single base station, and the support of legacy handsets. The disadvantages are the necessity for calibration measurements and database maintenance and potential capacity problems.

2.6.2 Digital TV (DTV) Location Technology

Rosum Corporation[6] has developed a positioning technology that uses digital television (DTV) signals. According to Rosum, the technology provides mobile users with meter-level accurate positioning indoors and in urban areas. The solution uses DTV signals to determine location. Each 6 MHz-wide signal is broadcast at powers up to one megawatt. These signals are in the upper UHF spectrum, a series of frequencies ideal for urban propagation. High bandwidth, high signal power, no modulated data, and known synchronization sequences can be found in these television signals that are being broadcast by stationary towers. Because of it high signal strength, the DTV positioning solution provides superior coverage indoors and in urban areas. The wide bandwidth DTV signal is particularly resistant to multi-path errors.

Receivers tracking these signals may operate without aiding data, without real-time Doppler information and in the presence of severe attenuation. Unlike GPS signals that are transmitting at around twenty watts from over a thousand kilometers above the earth's surface, DTV broadcasts at megawatt levels at distances of a few tens of kilometers from the user. DTV's 6 MHz bandwidth is six times that of GPS. Unlike the orbiting GPS satellites, DTV stations are stationary with respect to the mobile user. This simplifies the computation and eliminates the need for real-time Doppler and ephemeris[7] data. Because of the strength and high bandwidth of the signal, Rosum predicts that its technology will work much more dependably indoors than competing GPS solutions.

The standard TOA approach is used, that employs time-stamped pseudorange measurements from at least three DTV towers to compute the position. A

[6] http://www.rosum.com

[7] A table of predicted satellite orbital locations for specific time intervals.

monitor unit samples the available DTV broadcasts in an area and models the clock drift for each of the towers by measuring the drift against an atomic standard. The data generated by the monitor unit is sent to the position determining equipment and is used in the position calculations.

Another key advantage of the system is that it is not necessary to demodulate the actual DTV data. It is only necessary to find the synchronization codes in order to produce an accurate pseudorange. This provides over 50 dB of processing gain above that of a standard television receiver. In fact, the digital television signals can be used for ranging at distances of over 150 kilometers.

2.6.3 Ultra Wideband (UWB) Location Technology

Very narrow pulses, whose width is measured by several hundred picoseconds, characterize Ultra Wideband (UWB) technology. These short pulses spread through a very wide frequency band. The frequencies may overlap with bands that have been already allocated to other applications such as licensed wireless and GPS bands. To prevent interference with other devices operating across the UWB frequency range, the UWB signal level should fall below -41 dBm in that range.

UWB could be very beneficial for indoors positioning because the high bandwidth of the technology implies less susceptibility to multi-path effects. Multipath is quite pronounced inside buildings and reduce positioning accuracy. The narrow pulses in the time domain allows very high accuracy (500 picoseconds is equivalent to 15 centimeters).

Using UWB technology, the position of a transmitting mobile can be determined by the time difference of arrival (TDOA) of its signals at fixed position reference receivers. As we have seen, the TDOA method requires synchronization of all the reference receivers with respect to a common clock. In another approach, the position of the transmitter could be determined from the signal round-trip delay (RTD) to the reference receivers. The reference receivers would then retransmit the signals that enable the mobile that measure the RTD. The RTD approach does not require clock synchronization with the reference receivers. However, it does require the knowledge of the retransmit delays of the reference receivers, which has to be subtracted from the RTD.

A drawback for UWB technology is that it requires a reference infrastructure of receivers/transmitters. However, if these beacons would be used to access other applications or deployed in a context of a WLAN (Wireless Local Area Network), the cost of the infrastructure could be shared among the various applications.

Despite its low power output, concerns remain on the effects of UWB on navigation and positioning that depend on GPS technology. Specifically, it could reduce the A-GPS capability in an indoor environment. While there is a legitimate concern about possible interference of UWB with other RF technologies, UWB could have a positive effect on location technology. A hybrid technology that combines GPS with UWB may provide a solution that enables seamless, indoor, and outdoor positioning capabilities to location services.

2.6.4 Advanced Maximum-Likelihood Estimation

LOCUS Corporation[8] developed a new network-based positioning solution - the Advanced Maximum-Likelihood Estimation (AME). AME requires neither changes to handsets nor investment in base stations for TDMA (Time Division Multiple Access) based wireless radio interfaces such as IS-136 and GSM. Unlike location technologies (such as TDOA, EOTD and, A-GPS) AME does not require additional network-based measuring devices.

AME employs the MAHO[9] (Mobile Assisted Hand Off) procedure that is defined for TDMA IS-136 and other wireless radio interface standards. Using Mobile Assisted Hand Off, the system requests the handset to measure signal strength of up to 24 neighboring base stations on broadcast control channels. After the handset make the measurements of the MAHO signals, it transmits this information back to the wireless system.

The AME Positioning Determining Equipment (PDE) includes a database that stores profile information on grid points for the coverage area. The profile includes signal strength mean and standard deviation for the grid points. Field measurements as well as RF computer simulation combined with three-dimensional digital maps are used to populate the database.

Using the measured signal strength levels and standard deviations from multi-

[8] http://www.locus.ne.jp

[9] The MAHO procedure was designed to help the wireless system determine the best cell site to handoff.

ple base stations, the PDE computes maximum-likelihood equations for a grid point for which the probability for locating the handset is maximized.

2.7 Comparison of Accuracy

A very important parameter is accuracy of the positioning method, as it defines which applications can be provided. With today's widely deployed Cell-ID method, it is impossible to provide satisfactory navigation services. Especially in urban areas localisation is too inaccurate to determine on which street the user exactly is at the moment, not to mention the feedback time which is needed should the user have taken a wrong turn. As can easily be seen, accuracy of about a few meters is necessary. This means that higher accuracy offers more possibilities. All LBS applications that are being developed today (such as car navigation, localisation of lost items, persons or pets, precise location of emergency calls, and interactive games) will be complemented with new services once accuracy is increased. It cannot be the aim to foresee new and successful services, as we have stated before, it is also not possible to forecast acceptance of these services by the society. Again, however, we want to state the corollary of the "Field of Dreams Approach" - "If we do not build it, they surely cannot come".

A comparison with downhill skiing or formula one racing shows the importance of accuracy. Only exact time measurement in the range of one hundred of a second makes these sports so popular and interesting to the fans. Without exact time taking, these sports would probably be a lot less popular and a lot of other tournament sports might not even exist.

Of course, accuracy will not be the only factor to determine which and how many applications for LBS will be deployed. Other questions that need to be solved are sufficient and guaranteed QoS, tariff models, availability of mobile equipment for the method in question, security and privacy issues, and other questions that might be of importance to the user. Some of these questions have already been considered in the introduction. This part concludes with a presentation of the state of the art in achieved accuracy of the methods in question.

Method	General accuracy	Urban	Sub-urban	Rural	Indoor
Cell-ID	100 m – 35 km	328 m [Ekholm01]	639 m [Ekholm01]	2–35 km	No change unless there is a pico-cell
Cell-ID + TA	100 m – 35 km	283 m [Ekholm01]	415 m [Ekholm01]	2–35 km	No change unless there is a pico-cell
RXLEV	100 m – 10 km	207 m [Ekholm01]	448 m [Ekholm01]		Slight degradation but penetrates well indoors
TOA	40–150 m				Slight degradation because of multi-path but penetrates well indoors
E-OTD	50–150 m	125 m [Larder01]	50 m [Larder01]	50 m	Slight degradation but penetrates well indoors
GPS	20–100 m	20–100 m	20 m	10 m	Very bad
A-GPS	5–50 m	10–50 m	10 m	5 m	Better than GPS but still not good enough
AOA	50–150 m	125 m	45 m [Swales99]	50 m	Bad because of multi-path
DCM	50–150 m	44 m [Laitinen01a]	74 m [Laitinen01a]		Bad because of multi-path

Table 2.1: Methods accuracy

3 LBS Standardization

One of the main motivations for location-based services is to generate additional revenues for mobile operators, because of the high investments for UMTS. As we have seen in the introduction this refers to Europe in particular. In the USA there is additional momentum caused by a ruling of the Federal Communication Commission (FCC). The FCC[1] is the main American regulatory authority in the area of telecommunication. Among others the FCC administers and allocates the whole frequency spectrum on local, regional, and national level. Since 1996, the FCC has specified the enhancement of the 911 emergency call for mobile users. This new service is called E911[2] (Enhanced 911) and requested to be supported by all American mobile network operators.

Therefore, it became a rule set by the regulator, which obligates all mobile network operators to implement location techniques in their networks. The exact requirements will be presented beneath. At this point, it is important to point out the obligation to locate the mobile user with a higher accuracy than the one given by the cell-ID method.Thereby, the network operators are forced to implement better methods with higher accuracy, such as have been described in Chapter 2. In the mean time, provision of the requested accuracy has proven to be a very difficult technical challenge and there are also other technical and non-technical challenges. Thereby the original operational terms have been extended.

In Section 1.2.1, we already specified methods for reaching the threshold of the numbers of users. The FCC rules are not directly intended to support the evolution of LBS services but rather to raise the security of the mobile subscriber. Nevertheless, this will expedite the development in the LBS area. The

[1] http://www.fcc.gov

[2] http://www.fcc.gov/e911/

interests of mobile network operators are to develop other services alongside E911 because the cost of implementing new location methods in their networks will be very high. In this sense, there will be a crossing of the threshold of the number of users, leading to a mass market for LBS applications.

Location-Based Services as the basis of new revenues for the European network operators and the obligation for more security in American mobile networks are two main forces for standardization of location methods in GSM, GPRS, and UMTS. The entities in charge of the location service standards for these interfaces are the European Telecommunications Standards Institute (ETSI) and the Third Generation Partnership Project (3GPP) in Europe, and the T1P1.5 group of the American Standards Committee T1.

We have already discussed the importance of standardization in the introduction part. Standards should enable faster spreading of new technologies. LBS in mobile networks do not only affect the radio interface and location methods, but also network internal aspects such as architecture and protocols. This part of the evolution depends also on the category of LBS services.

In general, there are four categories of usage of the location service:

- **Commercial LBS** (or Value Added Services, VAS) – are services offered for the retail customer base. Without commercial LBS there is little chance for LBS to take off and reach the threshold of Metcalf's law. This law states that the usefulness of a service equals the square of the number of users.

- **Internal LBS** – will typically be developed to make use of the location information of the Mobile Equipment (ME) for access to network internal operations. This may include, for example, location assisted handover and traffic and coverage measurement. This may also include support for certain O&M related tasks, supplementary services, GSM bearer services, and teleservices.

- **Emergency LBS** – will typically be part of a service provided to assist subscribers who place emergency calls. In this service, the location of the mobile caller is provided to the emergency service provider to assist them in their response. This service may be mandatory in some jurisdictions (i.e., in USA).

- **Lawful Intercept LBS** – will use the location information to support various legally required or sanctioned services.

Common to all LBS applications and independent of the group affiliation, is the importance of a proper QoS, privacy, and availability.

In the first part of this Chapter, we will present the FCC rule and the European counterpart to the enhanced emergency call for mobile users. The second part defines the standardization for LBS in GSM, GPRS, and UMTS networks. In addition, standardization efforts of the WAP Forum, the Bluetooth SIG, and the Open Group Consortium (OGC) are part of this section. With an overview on different protocols in the Location Interoperability (LIF) Forum and the IETF we conclude this chapter.

3.1 American E911 Service

The US Federal Communication Commission (FCC) has made Enhanced 911 (E911) a mandatory requirement for wireless communications services such as cellular telephone, broadband personal communications (PCS), and geographic area specialized mobile radio (SMR). This ruling and upcoming service is called wireless E911. For Phase I implementation, the FCC required that Public Safety Answering Points (PSAP)[3] attendants of wireless communication networks must be able to know a 911 caller's phone number for return calls and the location of the base station or cell cite (Cell-ID) of the caller so that calls can be routed to an appropriate PSAP and related emergency assistance attendants. As part of Phase II, wireless carriers are required to provide Automatic Location Identification (ALI) by which the position has to be recorded in latitude and longitude. A certain position accuracy has also been defined, depending whether the solution is mobile- or network-based:

- For mobile-based solutions: 50 m in 67% of calls and 150 m in 95% of calls.
- For network-based solutions: 100 m in 67% of calls and 300 m in 95% of calls.

[3]Organizations that respond to emergency calls.

The choice of the location method is left to the mobile network operators. The precondition is that all operators will support ALI as a part of Phase II of E911.

In the year 2000, the FCC required wireless communication operators to offer operational location-capable phones by October 1, 2001. In September 2000, the FCC granted a limited waiver to VoiceStream with relaxed accuracy for an extended period. Right after the October deadline of 2001, waivers were granted to Alltel, AT&T Wireless, Cingular, NEXTEL, Sprint PCS, and Verizon, permitting them to postpone activating location-capable phones that satisfy the Phase II requirements until 2002 or later. Presently, the FCC established a four-year rollout schedule for Phase II, beginning October 1, 2001 and to be completed by December 31, 2005.

On the operators side it seems now that the focus has shifted increasingly to actual implementation. The initial discovery, the development and evaluation phase seems largely completed. This, in turn, has focused increased attention on implementation issues like PSAP readiness. It seems that the problem is that the existing 911 infrastructure is in no condition to accommodate the pervasive use of wireless technologies, the Internet, or the many other product offerings that invite or demand access to 911 services [Hatfield02]. The 911 network has been fundamentally unchanged since its rollout in the 1970s and remains a separate network burdened with point-to-point analog circuits, inband signaling and low-speed analog data lines [Corporation01].

On the other hand, in the course of time the requested positioning accuracy has shown to be a difficult prerequisite. At this time, the favored methods are A-GPS and E-OTD.

3.2 European E112 Service

The executive body of the European Union (EU), the European Commission (EC)[4], has similar initiatives for their wireless emergency calls, E112 (Enhanced 112). According to their definition:

> "E112 means an emergency communications service using the single European emergency call number, 112, which is enhanced with

[4]http://europa.eu.int/comm/index.htm

location information of the calling user."

Coordination groups within the EC have been organizing meetings to specify similar requirements as their counterpart in the United States. Looking at the American regulator experience, the EC has taken a very conservative attitude since end of 2001. First of all the directive 2002/22/EC on universal service and user rights requires public telephone network operators to make caller location information available to authorities handling emergencies: "to the extent technically feasible". In addition, during the introductory phase of E112 services, application of the best efforts principle for location determination is considered preferable. However, as Public Safety Answering Points (PSAP) and emergency services gain practical experiences with location information, their requirements will become more defined. Moreover, location technology will continue to evolve, within both mobile cellular networks and satellite location systems. Therefore, the best effort approach will need to be reviewed after the initial phase.

It was recommended by the EU communication review that location for emergency purposes should become mandatory by January 2004. However, the EC will be reviewing operator compliance with its E112 Directive and Recommendation in late 2004, and it is unlikely that location-based emergency calls will be able to perform as expected, due in part to the slower than expected rollout of commercial location-based services. The Commission will take stock of progress made and assess whether further action at EU level will be needed.

E112 is being investigated as part of the fifth frame program of the Information Society Technologies Programm (IST)[5] for technical research in the European union. There are more than 70 projects in relation to LBS aspects. Some of them have already been finished like LOCUS[6], Cello[7] and CGALIES[8] and have delivered very useful insights. A very good overview of all projects can be found at the POSITION site[9]. POSITION is an IST accompanying measure that aims to create a synergy between European positioning-related projects, through the cross-fertilisation of results.

[5] http://www.cordis.lu/ist/

[6] http://www.telematica.de/locus/

[7] http://www.telecom.ece.ntua.gr/cello/

[8] http://www.telematica.de/cgalies/

[9] http://position.fdc.fr/

3.3 GSM Standardization

In the GSM and UMTS standardization process the term Location-Based Services is used as the term for a service provided by either a network provider or a third-party service provider that utilizes the available location information of the terminal. The corresponding service concept in the GSM and UMTS system standardization is called LCS (Location Services). Thereby LCS specifies all necessary network elements and entities, their functionalities, interfaces, and communication messages necessary to implement the location service functionality in a cellular network. In the following sections about standardization in GSM, GPRS, and UMTS, we shall stay compliant according to the standard and therefore we will use the term LCS wherever appropriate.

As we have already seen, GSM standardization activities for location services cover both the location techniques and the corresponding network infrastructure. For the GSM interface, all LBS standards are mature. The standard bodies do not expect any more releases in the near future. Three approaches for location determination are included in GSM standardization:

- Time-of-Arrival (TOA),
- Enhanced Observed Time Difference (E-OTD), and
- Assisted Global Positioning System (A-GPS).

In addition, the standard supports usage of Cell ID and Timing Advance in the case that the network and/or the terminals do not have the required functionalities for the more sophisticated methods. The mentioned techniques have already been described in Chapter 2. In the following, the architecture supporting location is described. The generic LCS logical architecture is shown in Figure 3.1 [3GPP04b].

Location services in GSM are enabled by a number of network elements and other entities referred to as LCS components. The whole system consisting of distributed components is referred to as LCS Server. An external client is an entity that interacts with the LCS Server via the L_e interface. Thus, it can also be a third part application. The L_e is only stated as a reference point and not as an interface. If defined, L_e can allow roaming between different PLMNs. Clients subscribe to the appropriate service in order to obtain location information of mobile terminals. In most cases, the client is an external meaning such that the client is not part of the GSM network.

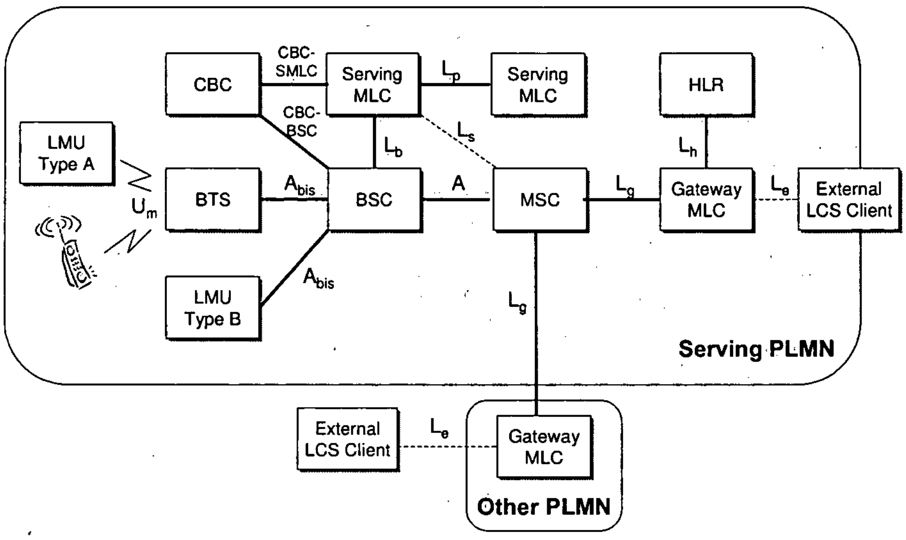

Figure 3.1: GSM network LBS architecture

The external client connects to the server via the GMLC (Gateway Mobile Location Center). After performing registration and authorization of the client, the Gateway-MLC forwards the positioning requests to the VMSC (Visited Mobile Switching Center). The Visited-MSC performs a number of essential tasks to find out the location of the mobile, for example:

- Checks the subscription authorization of the Mobile Subscriber (MS) to be located,
- Retrieves the information of which location method to use based on the mobile subscriber classmark and the network capabilities,
- Initiates required procedures depending on the location method,
- Coordinates the flow of incoming location requests, and
- Interfaces with functions of Serving-MLC in order to get the location estimation of the mobile station.

The mobile station classmark is sent at the beginning of a connection and it indicates which one of the following implementation options is used:

- Location Class A (LC-A) – The Position Calculation Function (PCF) is in the mobile station and does not require assistance data from the infrastructure whatever the requested accuracy.

- Location Class B (LC-B) – The Position Calculation Function is in the mobile station and requires assistance data from the infrastructure at least for some requested accuracy.
- Location Class C (LC-C) – There is no Position Calculation Function in the mobile station.
- Location Class D (LC-D) – Location functions are not supported (e.g., legacy mobile subscribers).

In the case of location class C, the Serving-MLC coordinates and carries out the actual positioning function. There are several Serving-MLCs in the network connected either to the Mobile Switching Centers or the Base Station Controllers (BSC). Due to this, the support for Serving-MLC can be either Network Subsystem (NSS) based or Base Station Subsystem (BSS) based, which is the case in GSM only. As we will see later, UMTS specifications support only the Base Station Subsystem based variant of Serving-MLC implementation. The network based Serving-MLC is connected to the Mobile Switching Center, and the base station based SMLC is connected to the Base Station Controller. Probably both, network- and base station-based SMLC, can exist in parallel in one network, but the practical implementation of such a network configuration may be demanding. Furthermore, an assumption in GSM LCS stage 2 recommends to standardize a similar open interface to the Serving-MLC whether it is based on the network or on the base station. This simplifies migration from an network- to a base-station-based location architecture and avoids two different types of Serving-MLC. In addition, another assumption in GSM LCS stage 2 states to enable migration from a network-based SMLC to base-station-based Serving-MLCs. Therefore, even GSM LCS stage 2 supports the trend toward utilizing SMLC based on a Base Station Subsystem.

The Serving-MLC retrieves information required by the location algorithm from the LMUs (Location Measurement Unit). There are two types of Location Measurement Units. From the functional point of view Type-A LMU is similar to a mobile terminal. It is located at a known location, typically at a base station site, where it carries out time delay measurements towards base stations. In this way, it provides reference measurements for the E-OTD method. The Serving-MLC communicates with a Type-A LMU through the standard GSM radio interface. Type-B LMU carries out measurements for the TOA method and is accessed via the A_{bis} interface. In addition to LMU data, the Serving-MLC collects the required signal measurement results from uplink and downlink as well as retrieves the geographic coordinates of the base stations. Using all this information, the Serving-MLC performs location calculations and converts the coordinate estimation to the preferred geodetic reference system. The estimated coordinates of the user are then sent to the client via the visited MSC and the mobile location center gateway.

The Serving-MLC may use the services of the Cell Broadcast Center (CBC) in order to broadcast assistance information to several users at the same time. This can be information required for A-GPS positioning or terminal-based E-OTD positioning. Additionally, it handles the ciphering of location services assistance data for the assistance data broadcast function.

The GSM standard defines a generic Radio Resource LCS Protocol (RRLP) that is used to transfer location services related information between the mobile station and the Serving-MLC. The positioning flow and the corresponding activities are given below.

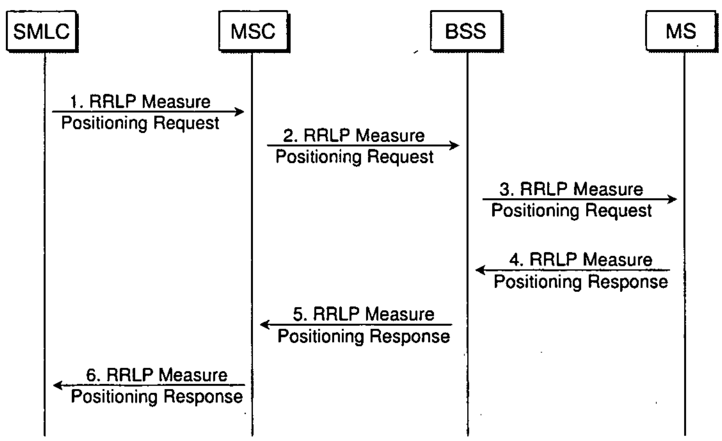

Figure 3.2: GSM positioning flow

1. The Serving-MLC determines possible assistance data and sends it to the MSC.

2. The MSC forwards it to the Base Station Subsystem (BSS); respectively the Base Station Controller (BSC).

3. The BSC sends the positioning request including the QoS and any assistance data over the base station to the mobile station.

4. In case of a mobile-based method (like E-OTD), any data necessary to perform these operations will be either provided in the RRLP Measure Position Request or made available from broadcast sources. There is a need for assistance data to be broadcast to the mobile station. Here, we differentiate two cases:

   - Mobile-based – contains the Real Time Difference (RTD) values (in case of a non-synchronised network) and base station coordinates. In addition, the broadcast data contains other information simplifying the positioning measurements. The typical size of one broadcast message is less than 82 octets.
   - Mobile-assisted – contains identities of base stations to be measured, what channels they use, etc. The resulting positioning measurements (MS-assisted) or location estimate (MS-based) are returned to the BSC in a RRLP Measure Position Response.

   In the case of mobile-based the uplink will contain less flow than mobile-assisted.

5. The BSC sends measurement results within BSSMAP Location Information Report message to the MSC.

6. The MSC forwards the measurement results within LCS Information Report message to the Serving-MLC.

Concerning privacy, a Privacy Override Indicator (POI) is used to determine whether the privacy settings of the subscriber to be positioned shall be overridden by the request for location services (lawful intercept). The type of LCS client requesting location information (i.e., commercial, internal, emergency, or law-intercept) shall determine the value of the POI assigned to the LBS client profile.

3.4 GPRS Standardization

The importance of standardizing Location Services in General Packet Radio Service (GPRS) relates to the support of location based services in a packet-switched architecture. These standards are enclosed in the 3GPP GERAN (GSM EDGE Radio Access Network) LCS Release 5 [3GPP02]. The prerequisite was to keep changes to GPRS to a minimum.

As can be seen in Figure 3.3, there are new interfaces between the Serving GPRS Support Node (SGSN) and the Gateway-MLC. The rest remains to be equal to the GSM architecture for LCS.

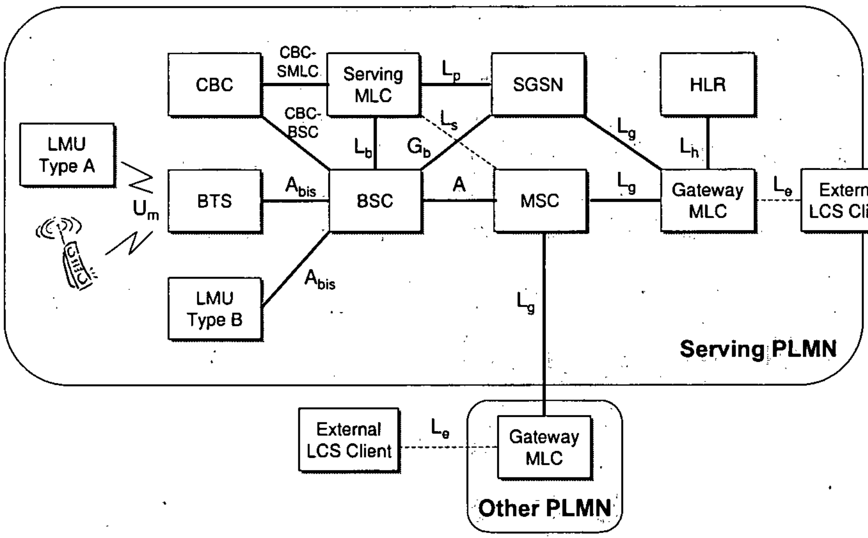

Figure 3.3: GPRS network LBS architecture

In GPRS three methods for LCS should be supported:

- Timing Advance (TA),
- Enhanced Observed Time Difference (E-OTD), and
- Assisted Global Positioning System (A-GPS).

Besides the standardization of interfaces and the transmission of LCS data over packet channels, work has been done on different service aspects. Among them are velocity estimation, personal security control, localization of all terminals inside a specific area (Location of All Mobiles in Geographical Area, LAMGA), and the establishment of defined geographical areas where specific services can be used (DEfined Geographical Areas, DEGA).

The Serving GPRS Support Node obtains the position of a Mobile Station by requiring it from the BSS (Figure 3.4).

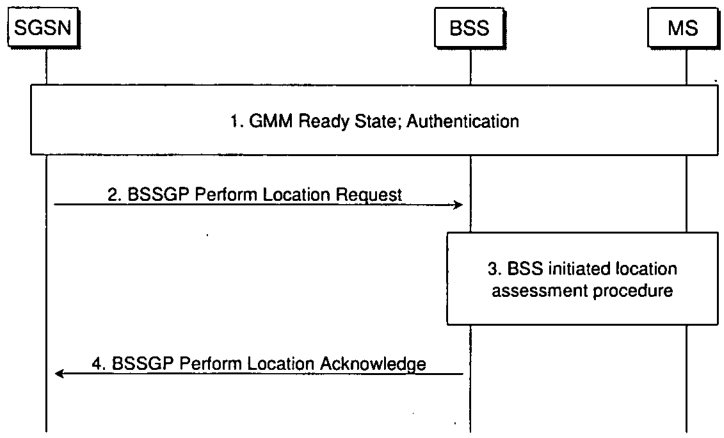

Figure 3.4: GPRS positioning flow

1. The procedure can be started whenever the Serving GPRS Support Node considers that the mobile station is in Ready State (i.e., the GMM Ready timer is running on the SGSN side). The Serving GPRS Support Node verifies first barring restrictions in the mobile user's subscription profile. If an application requires the location of a mobile subscriber for which the Ready timer is not running, the SGSN first pages the mobile subscriber. Authentication may be applied as required by the operator.

2. The Serving GPRS Support Node sends a BSSGP (Base Station System GPRS Protocol) Perform Location Request message to the BSS. This message carries the TLLI (Temporary Logical Link Identity) of the mobile station to localize and the QoS information.

3. The Base Station Subsystem runs location acquisition procedure. The procedures to apply are chosen according to the Base Station Subsystem and the mobile subscriber capabilities, as known from the mobile station classmark. The different cases have been described in Section 3.3.

4. The Base Station Subsystem answers with a *Perform Location Acknowledge* message with the best location estimate it can reach within the time ascribed by the QoS information as indicated in the *Perform Location Request* messages.

5. The Serving GPRS Support Node may record billing information.

3.5 UMTS Standardization

Different positioning solutions are being studied widely by network operators and research institutes. Currently, there are three 3GPP standardized location techniques supported by the UMTS Terrestrial Radio Access Network (UTRAN):

- Cell-ID,
- Observed Time Difference of Arrival (OTDOA) with optional Idle Period Downlink (IPDL), and
- Assisted Global Positioning System (A-GPS).

All of these methods have been described in Chapter 2. In the following part, we will review the client server architecture and operations within the UTRAN system standard based on [3GPP04c]. This architecture is based on a client server model with the mobile units carrying out measurements that are reported to a calculation server located within the network (e.g., the mobile-assisted method).

3.5.1 LCS Architecture in UMTS

Figure 3.5 illustrates the LCS architecture in UMTS release 6, as specified in the section "Network architecture" of [3GPP04a]. The figure indicates that Serving-MLC functionality of UTRAN is integrated in Serving-RNC. It is also noted in the specification that the usage of CBC for LCS assistance data in UMTS is for further study.

One main principle in 3GPP is to keep the radio network aspects purely in the access network specifications, and therefore it was not feasible to include Network Subsystem (NSS) based Serving-MLC in 3G specifications. The tasks of Serving-MLC are directly related to the radio access network. The I_u interface

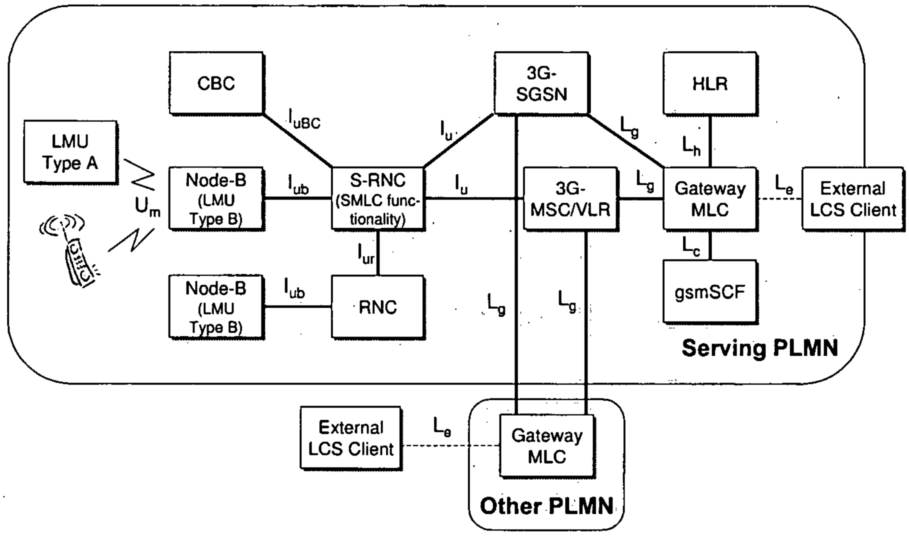

Figure 3.5: UMTS network LCS architecture

between the core network and the access network carries only the location requests and corresponding location reports. Support for network based SMLC was necessary in GSM because LCS support was built on top of the existing GSM standard. The base station based Serving-MLC in GSM was introduced since heavy signaling load on the A interface with network based Serving-MLC was anticipated. The signaling load on the I_u interface would also be substantial in 3G with a network based Serving-MLC.

Furthermore, if the Network Subsystem-SMLC is used only to generate and distribute assistance data for GPS, several such Serving-MLCs are needed in a network, since the GPS assistance data is only valid over a few 100 kilometers. Broadcasting GPS assistance data is demanding for timing accuracy reasons. In UTRAN, assistance data is included in system information blocks. A Cell Broadcast Center (CBC) could in principle be used to broadcast GPS assistance data also in 3G systems, but normally CBC needs not be involved at all. The Radio Network Controller collects and calculates the assistance data and includes it in the system information to be broadcasted in the relevant cell. As an alternative, the RNC could transfer the cell related assistance data to the Cell Broadcasting Center, which in turn would transfer the assistance data back to the RNC to be cell broadcasted. This complexity is the reason why the broadcasting of assistance data has been stated "for further study" in the network architecture specification [3GPP04a].

It is apparent that multiple options exist in the GSM LCS specifications, but these options have only added to the obscurity of the specifications and created additional difficulties in implementation. Adding too many options to a specification reduces the simplicity and effectiveness of one complete standard. The goal for LCS in UTRAN was, and still is, to optimize functionality and system architecture: it is easier from the standardization and signaling point of view to integrate LCS functionality in Radio Network Controller. RNC already has ready access to much of the information needed for location determination of a target user equipment. A network-based SMLC in UMTS would only increase system and standardization complexity in 3G.

LCS interfaces in the core network

The Gateway Mobile Location Center (GMLC) is the gateway between the client and the mobile network. GMLC is connected over standardized interfaces to the Home Location Register (HLR) (L_h interface), Mobile Switching Center (MSC) (L_g interface), Serving GPRS Support Node (SGSN) (L_g interface) and the L_c interface to Camel applications. The current understanding is that the L_e interface between the gateway and the client should be part of OSA (Open Services Access). The Location Inter-operability Forum (LIF) is standardizing support for location services that is also applicable for the L_e interface of gateways. The gateway interfaces are the same for 2G and 3G and possible enhancements are applicable for both 2G and 3G. Hence, there is evolutionary support of LCS interfaces provided in the core network.

3.5.2 UTRAN Location Architecture

The 3G UTRAN location service logical architecture and the location service elements are illustrated in Figure 3.6. The main elements for the Location Service include:

- LCS Client Function (LCF),
- LCS Server Control Function (LSCF),
- Position Radio Co-ordination Function (PRCF),

- Position System Measurement Function (PSMF), and
- Position Calculation Function (PCF).

The LCS System Operation Function (LSOF) provides storage of database information that may be used as part of the location process, network operation, and maintenance.

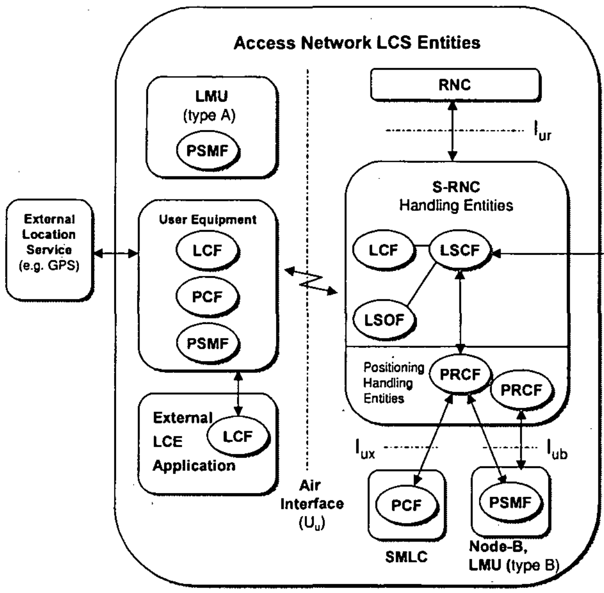

Figure 3.6: Location service architecture in UTRAN

Measurement and database resources across the network are combined to determine the geographic location of the user equipment (UE). External location services such as GPS may be used within the network as part of the location method (e.g., A-GPS). These are assumed as part of the calculation or database functions.

The request for the location of the mobile originates with an application program or LCS Client Function (LCF). The application may be a part of the user equipment (UE), in the mobile network, or in an attached network (e.g., the Internet). Location requests may also originate within processes inside the access network (e.g., location assisted handover). The location information request is forwarded to the LCS Server Control Function (LSCF) in the access network. In the UTRAN model, this entity is associated with the Serving Radio Network Controller (S-RNC) for the user equipment. The server coordinates, within the access network, the operation of the various service elements and radio equipment. The radio equipment has to measure the geographic location of the user equipment and to return the estimate to the client function.

The server requests the necessary measurement to be made by Position System Measurement Functions (PSMF) that may be located within the base station (Type-B), stand-alone (Type-A), or within the user equipment itself. The needed reference information is obtained from databases such as the Location System Operations Function (LSOF). This may include any necessary assistance data (e.g., GPS visibility data or calibrations, and the geographical location of the base stations), which is passed to the measurement units or to the user equipment. The necessary calculations are performed on the measurement and the data by the Position Calculation Function (PCF) positioned in the user equipment (mobile-based) or the Serving-MLC (mobile-assisted). This calculation includes not only the estimate of the location of the mobile but also the estimated error in the response and any needed coordinate transformations.

This logical architecture has been designed to be independent of the details of the location technology. The location method may involve measurements made by the mobile station (e.g., Time-of-Arrival), by the base station (e.g., Angle of Arrival), by Location Measurement Units (e.g., Observed Time Difference of Arrival) or involve an external location service (e.g., A-GPS). The selection of the location method for an individual request is carried out by the control function in view of the needed accuracy of the client function, the available network and mobile station resources, and the radio propagation and environment conditions. The calculation may combine information from several sources to achieve a more accurate location estimate. Various aspects of the location method technology may be easily upgraded as the service requirements evolve and technology advances. These changes may be introduced without disruption of operation or changes in existing LCS Client Functions and applications.

This architecture arrangement has also been designed to isolate the core mobile network from details of the radio access network so that common LCS applications in the mobile network can use different access networks. Within the core network are the detailed subscriber functions involving service subscriptions, privacy rules, accounting, and charging processes. Within the access network are the location functions and the radio resources management functions. By isolating these two groups of functions, independence of the general location service from the details of the technology can be preserved.

In this architecture, the control function is associated with the Serving-RNC. The Serving-RNC acts as the radio base station controller logically associated with the mobile station. The Serving-RNC is thus best able to manage the details of the radio communication and the environment associated with the mobile station for radio signal measurements and communications. These tasks are delegated to the Position Radio Coordination Function (PRCF), which manages the positioning for a mobile station through overall coordination and scheduling of resources to perform positioning measurements. The Position Radio Coordination Function determines the positioning method to be used based on the user equipment Positioning request, the QoS, the capabilities of the UTRAN, and the User Equipment's capabilities. The control function also manages the needed radio resources through the Position Radio Resource Management (PRRM). It determines which measurement functions are to be involved, what to measure, and obtains processed signal measurements from the same.

This logical association may mean that the control and the coordination functions are implemented as a part of the Serving-RNC or they may alternatively be implemented as a separate server unit that services a group of RNC's. This choice may be made based on traffic volumes and signaling communications.

The Serving-RNC responsible for the mobile terminal may change over time as the mobile moves about the coverage region. As it moves, the mobile terminal may become associated with new Serving-RNC as a normal part of the access network operations. The newly associated Serving-RNC will assume responsibility for LCS operations. If a request for location information is received while a handover is in progress between RNCs, then it is expected that the request will be deferred until the handover has been completed and the new Serving-RNC is able to service the LCS request.

3.5.3 Signaling and Privacy

Message signaling, standard to the access network, is used to communicate among the LCS clients, the user equipment, the Serving-RNC, the LCS server, and the measurement units. The general service operation between the core network and the access network is illustrated in Figure 3.7. It outlines the basis for operation and interaction with the core network and LCS Client Functions.

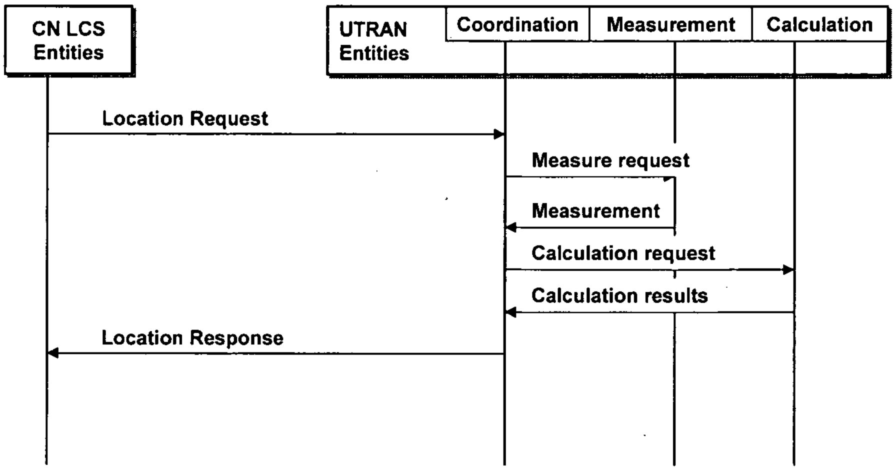

Figure 3.7: General UTRAN positioning flow

Upon receiving an authorized request from an LCS Client or for network internal operations, the LSCF will:

- Request measurements, typically from the mobile terminal and one or more base station radio units,
- Receive the measurement results,
- Send the measurement results and data to the appropriate Position Calculation Function,
- Receive the result from the Position Calculation Function, and
- Send the results to the LCS Client or to application clients within the access network.

In the event that the client is internal to the access network, the location information request may be made directly to the control function as internal clients are considered to be "pre-authorized". Some clients may require for their applications a more rapid response than others. The server, calculation functions, and signaling communications must be engineered as part of the network implementation to assure the timely response to location information requests.

As part of its operation, the location calculation may require additional information (e.g., base station coordinates). This may be obtained directly by communication with a database, or it may he through a request to the control function that will mediate the request and return of information from the appropriate database.

There may possibly also be available independent information that is able to supply the location information directly, or may be able to supply auxiliary information to the calculation function. The control function, as part of its activity to supervise the location process, may query the user equipment or other elements of the access network to determine their capabilities and use this information to select the mode of operation.

An important consideration in the location service is the privacy and confidentiality of the location information. Subscribers to the service may wish to restrict the availability of their location information to trusted clients or applications of their own choosing. The operation of the service thus begins with the client request being authenticated and authorized by an authorization center within the core mobile network. This service authenticates the client request, the subscriber subscription and privacy rules and authorizes the LCS information request. If there are charges associated with the location service, then the appropriate accounts are recorded. These subscription, charging, and service functions are independent of the access network and the radio technology. The authorization center may query the user equipment or other databases (e.g., HLR) to obtain the needed privacy and subscription information. This may involve message signalling that is standard to the radio interface. The authorized request is then passed to the control function in the S-RNC for the client.

Figure 3.8 illustrates in more detail the operations when the authorized request for location information is received by the control function[Steer00]. As many of these operations are internal to the Radio Network Controller, this diagram is presented to illustrate the logical information flow and implementation may use alternate arrangements. Not illustrated is the signaling used to initiate the location service request from the Location Client Function or from a mobile terming or Internet-based application.

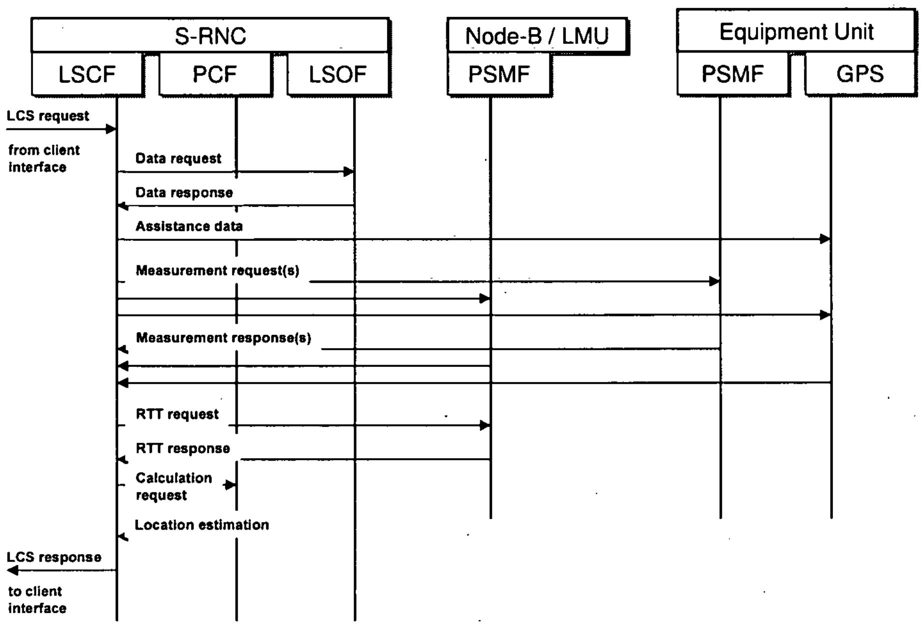

Figure 3.8: UTRAN positioning flow

1. The location operation begins with an authenticated request for location information about a user equipment being received at the control function (LSCF) from an application (client function), typically in the core network. The control function acts as the interface between the client functions in the core or the access networks and the location entities in the access network.

2. The control function considers the request and the capabilities of the mobile terminal and the network and selects the appropriate location method to be used for this request.

3. The control function requests from the database (LSOF) any information needed for the location method. This may include base station coordinates, information such as GPS satellite visibility, the relative time difference (RTD) between base station transmissions, or other calibration

information. The database may thus invoke measurements to maintain the calibration and real-time availability of the location system.

4. The control function directs the sending of any assistance data needed by the location method to the mobile terminal using the system standard broadcast or individual messaging channels. If the assistance data (such as GPS availability) is encrypted as part of the subscription process, the user equipment may receive information through a message exchange with the core network.

5. The control function initiates requests for measurements from the entities appropriate to the method being used. This may include measurements by the base stations involved with the user equipment from LMUs, from the mobile terminal, or from a combination of these.

6. The control function receives the measurement responses from the measuring entities. If the location method being used involves the use of an independent location facility, such as GPS, then the response may include a location estimate. This may be communicated directly to the client function (step 11) or it may be used to supplement calculation involving other measures (step 9).

7. Depending on the method chosen and the availability of requested measurements, the control function may request a measure of the round-trip-time (of the radio signals from the base station to the mobile) from the Position System Measurement Function (PSMF) in the serving base station.

8. The PSMF in the serving base station returns the round-trip-time measures to the control function if they were requested.

9. The control function passes the measurements, the data base information, calibration data, and, if necessary, round-trip-time information to the Position Calculation Function (PCF). It further requests a location calculation. The calculation request may include a coordinate transformation to the geographic system requested by the application.

10. The calculation function returns the location estimate to the control function. This estimate includes the location, the estimated accuracy of the results, and the time of day of the estimate.

11. The control function passes the location estimate to the requesting client function.

3.6 WAP Forum

The WAP Location Drafting Committee (LocDC) started work on location specifications in January 2000. At that time, there were groups defining location application programming interfaces (APIs) in the context of the intelligent network (IN), e.g., Parlay. However, no other groups were addressing location for Internet services [Adams03].

In February 2001, the WAP Forum received a draft of the Mobile Location Protocol (MLP) from the Location Interoperability Forum (LIF), as a WAP members contribution. The LIF MLP 1.0 specification was used as basis for the current data structure and services that the WAP location interface has today. The location specifications consist of [Forum01]:

- Location framework,
- Location protocols, and
- Location XML document format.

For the WAP Forum, location-based WAP services represent a class of applications with specific needs. The WAP location framework addresses these needs by providing a transparent as well as a position-procedure independent location application interface. At present, it defines three services - query, attachment, and external interface functionality to communicate, request, and reply to a location query using XML document exchange on HTTP (Hypertext Transfer Protocol), WSP (Wireless Session Protocol), and WAP PUSH.

- Location Query Services – allow an application to query the WAP location query functionality for the location of a WAP client. If there is only one request and response it is defined as the Immediate Query Service . An example application is a tracking application (e.g., fleet management) that wants to track WAP clients by request. If this application wants to track the WAP clients periodically then the service is called Deferred Query Service because of multiple deferred responses on one request.

- Location Attachment Service – attaches location information to a WAP client request. An example application that could use the location query service is the kind of application such as "Where is the nearest ...?".

- External Interface Functionality Service – allows a WAP client to request the location via an external functional interface in the terminal. An example application is a WAP client-based navigation application.

XML-based technology is used to represent and transfer location information, due to the speed with which it can be developed and its capability for richer data, enhancing the first 3G location-based services. The content is relatively simple because issues such as privacy, context, personalization, and positioning technologies were considered out of the scope and separate technologies are being created to deal with presence.

Since June 2002, the WAP Forum has been reorganized into the Open Mobile Alliance (OMA). The formation of the OMA has helped the move towards convergence with LIF via a MoU (Memorandum of Understanding). It will lead to a solution of the divergent standards problem in the short/medium term, with harmonization work already began.

3.7 Bluetooth SIG

In February 2000, the Local Positioning Workgroup was founded by the Bluetooth Special Interest Group (SIG)[10]. The main goals are to describe how to use Bluetooth for location services, to define the main application settings, and to resolve their influence upon technical specification of Bluetooth.

The goal is interoperability of devices for positioning services. On the one hand, positioning information should be exchanged between Bluetooth terminals and, on the other hand, the local positioning has to be compatible with other location methods such as GPS, A-GPS, and network-based methods. The Local Positioning Workgroup identifies the following five application settings:

- The Bluetooth device determines its own position by one or more devices in his environment.

[10]http://www.bluetooth.com/sig/sig.asp

- Devices learn and memorize their position. If two Bluetooth devices reach each other and only one of them is capable of positioning, also the other will obtain the positioning information. By this, the device will calibrate its infrastructure. The position will be memorized only during the time these two devices are in reach and can be handed over to third device in reach.
- Adaptive interpolation between devices. Inside a building there are more Bluetooth devices whereby the positioning information for all is acquired by only a few devices and the individual position of all is interpolated.
- Power saving. If a GPS device comes inside a Bluetooth environment, positioning information will be exchanged and the GPS device will switch to stand-by. When changing the position there will be an update over the Bluetooth devices.
- Using Bluetooth devices together with a GPS receiver inside a car. If GPS signals cannot be received, a Bluetooth device can apply cellular positioning methods and display the positioning information on the GPS display in the car. For real-time and tracking applications, GPS positioning information can be send to the mobile phone via Bluetooth. In an emergency situation (e.g., by activation of the airbag) the mobile phone can setup a connection and hand over the GPS information.

3.8 Open GIS Consortium

The Open GIS Consortium (OGC)[11] is an international industry consortium of 256 companies, government agencies and universities participating in a consensus process to develop publicly available interface specifications. GIS stands for *Geographic Information Systems*. In October 2000, the OGC started the OpenLS initiative (OpenLS^{TM} Initiative)[12]. The goal is the development of interfaces specifications that simplify the use of spatial and local information in a wireless Internet environment. The interfaces should be application, network, user-classes, and product-classes independent. The first step was the development of the OpenLS Testbed, which ended in January 2004 with a successful

[11] http://www.opengis.org/
[12] http://www.openls.org/

multi-vendor, specification-based mobile demonstration of these interfaces in action.

Many companies comprise the LBS market world-wide, but most of their interlinked LBS business opportunities await consistent communication of location and time, route, types of service, etc. across different providers, technology platforms, application domains, classes of products, and national regions. Markets will materialize when vendors can build a rich profusion of commercial brands and proprietary products and services on top of an integrated standards infrastructure. OGC's OpenLS Testbed produced much of this standards infrastructure, including the primary deliverables, two Open GIS implementation specifications and six interoperability program reports.

On the other hand, OGC accelerates the development of standards with special interest in gateway access to local content of the Internet. OpenLS continuing their work in the following areas:

- Location services and location-based content,
- Gateway services for the integration of location services together with mobile devices, wireless platforms, Internet protocol platforms and/or mobile location systems used between wireless IP systems and location-based services, and
- OGC service framework or service and content protocol as developed and tested in the OGC Testbed or other initiatives.

The intent of OGC is also to establish the harmonization of OpenLS specifications together with other industry forums and standardization groups like the Open Mobile Alliance (OMA) and particularly with their Location Interoperability Forum (LIF) working group.

3.9 Location Interoperability Forum

In October 2000, the world's three largest mobile phone manufacturers (Ericsson, Motorola, and Nokia) founded the Location Interoperability Forum (LIF)[13] to achieve the goal of offering location-based services worldwide on

[13] http://www.locationforum.org

wireless networks and terminals. The forum is dedicated to developing global interoperability between mobile positioning systems. The aim of LIF is to produce a common view on positioning technologies and system solutions to meet the emerging service requirements such as information retrieval and mobile commerce applications. LIF is therefore specifically focused on the user rather than on the operator perspective. In other words, this forum aims at identifying technologies and opportunities to provide location-aware services and not to use such data to enhance network management policies. LIF also handles security issues related to user's location data. The first applications based on the LIF recommendations were expected to be available already starting in 2001. LIF's purpose is to define, develop, and promote a common and ubiquitous location services solution, through the global standard bodies and specification organizations. Such a solution will:

- Define a simple and secure access method that allows user appliances and Internet applications to access location information from the wireless networks irrespective of their underlying radio interface technologies and positioning methods.
- Promote a family of standards-based location determination methods and their supporting architectures, which are based on Cell-ID and Timing Advance, E-OTD (GSM), AFLT (IS-95), and MS-Based Assisted GPS.
- Establish a framework for influencing the global standard bodies and specification organizations to define common methods and procedures for testing and certifying the LIF-recommended access method and positioning technologies.

LIF's intent is to have representation from a mix of network operators, equipment manufacturers, and service providers responsible for deploying equipment utilizing this solution. Its members will define this solution and submit it to the working standards groups. The members will then support the solution developed in the LIF in the appropriate existing global standard bodies and specification organizations and in the deployment of their systems and services. It is worthwhile mentioning that LIF is not a standardization body but rather a forum whose objective is to influence standards to promote the development of location-based services.

Since 2002, the Location Interoperability Forum has consolidated their work into the Open Mobile Alliance (OMA), and no longer exists as an independent organization.

3.10 Open Mobile Alliance

The OMA had its first meeting in June 2002 and was attended by more than 200 key organizations in the mobile industry. There had been a growing feeling that the mobility applications industry was becoming far too fragmented to be successful, and that future co-operation between what had been different interest groups was the best way forward to ensure good future market growth for 3G mobile. There seems to be little scope for small-scale individual initiatives that are not coordinated within the 3G mobile industry as a whole. Competition within the industry is clearly required for success. However, this is best achieved through other differentiators once a common framework of standards for areas like LBS and mobile commerce has been agreed upon. The charter for the Open Mobile Alliance is to:

- Deliver responsive and high-quality open standards and specifications based upon market and customer requirements,
- Establish centers of excellence for best practices and conduct interpretability testing, including multi-standard interoperability to ensure seamless user experience,
- Create and promote a common industry view on an architectural framework, and
- Be the catalyst for the consolidation of standards forums and work in conjunction with other existing standards organizations and groups such as IETF, 3GPP, 3GPP2, and the W3C.

The goal of OMA will be collecting market requirements and defining specifications designed to remove barriers to interoperability. It should also accelerate the development and adoption of a variety of new and enhanced mobile information, communication, and entertainment services and applications. The definition of these common specifications and the testing of interoperability, should promote competition through innovation and differentiation, while safeguarding the interoperability of mobile services throughout the entire value chain, across markets, terminals, and operators.

The work previously handled within the WAP Forum is continued within the new OMA. At their annual general meeting in June 2002, the WAP Forum members ratified changes to their articles in order to fully support the creation of OMA. Additionally, the Location Interoperability Forum (LIF), the MMS Interoperability Group (MMS-IOP), SyncML Initiative Ltd, and the Wireless Village initiative have announced their intent to work with the OMA through signing memoranda of understanding (MoU) [Adams03]. Since 2004, the Mobile Gaming Interoperability Forum (MGIF) and the Mobile Wireless Internet Forum (MWIF) are also consolidated into OMA. Each group is forming a working group. For the development of LBS, the inclusions of the LIF is of particular significance because this incorporates the strong vendor support from Ericsson and Nokia that LIF have, and brings two diverging standards together.

The strategic importance of the OMA in the development of a wide range of 3G applications services (of course including LBS) over the next few years looks very high.

3.11 Internet Engineering Task Force

Location services are supported in cellular mobile networks but also local networks such as LANs and WLANs will include mechanisms to support it. Therefore, the Internet Engineering Task Force (IETF)[14] works on protocols which enable the realization of LBS through the IP layer.

Begin 2000, the IETF started the Spatial Location Protocol (SLoP) activity in the IETF working group 16. The objective of the activity was to specify a common protocol (implying also a common API) for obtaining location information in the Internet. This means that different location sources, devices, applications, etc., connected to the Internet would have one common way of communicating location information. Initially, the work considered a general location architecture where location information of locatable objects would be

[14] http://www.ietf.org

accessible on SLoP-servers in the Internet. In addition, the object could roam between SLoP-servers and there would be a server discovery mechanism for locating the server providing the location of a certain object. The protocol considerations included:

- A common data set to express location information,
- A framework for representing other location information expressions,
- Negotiation mechanisms for agreeing on what location information representation to use,
- Security mechanisms for securing the location information data and controlling who can access the information,
- Policy mechanisms for setting privacy policies for who can access what location information,
- Transmission and reliability mechanisms for the protocol, and
- Coding of the protocol messages.

SLoP servers should be reachable over any TCP/IP platform and concerning the common data set to express location information the current proposal is to use:

- Latitude,
- Longitude,
- Altitude,
- Time, and
- Accuracy.

The working group proposal was later steered by IESG (Internet Engineering Steering Group) to first focus on the protocol messages. This includes the definition of a default location data set, a framework for representing other location expressions, and other data needed in the messages. In addition, the task is also to identify and define appropriate policy and security mechanisms, as well as to check what existing protocols could be used for transferring the data.

The lack of support for solving the security issue was the main reason why the working pace in this group slowed down in 2001. Therefore in May 2001, the geopriv working group (geopriv-charter)[15] was established. The focus of the work of this group is authentication, authorization, nondisclosure, and integrity of location-based data. End of the work was scheduled for November

[15] http://www.ietf.org/html.charters/geopriv-charter.html

2004. Currently, there are three RFCs with privacy requirements, threat analysis of the geopriv protocol and DHCP option for coordinate-based location configuration information (RFC 3693, RFC 3694, and RFC 3825, respectively).

4 QoS Architectures

Providing quality to a specific service is the most important factor in term of making the service successful. Thereby it is not surprising that standardization organizations and many scientific and industrial activities are directed on Quality-of-Service (QoS) topics. This chapter is focused on QoS architectures of the transport network, based on work done in the IETF.

As we have defined in the introduction, there are three notions of QoS - intrinsic, perceived, and assessed. Intrinsic QoS pertains to service features stemming from technical aspects and it is the responsibility of a network provider and depends on network architecture, planning and management. Perceived QoS reflects the customer experience of using a particular service. Ensuring a high level of perceived QoS can be achieved by the appropriate use of the intrinsic QoS capabilities adjusted to a particular service offered. This is the obligation of the service provider. The assessed QoS starts to be seen when the number of users is passing the threshold of acceptance defined in Section 1.2.

ITU and ETSI approaches to QoS-related terminology are almost the same and adhere mainly to perceived rather than intrinsic QoS. On the other hand, IETF focuses on intrinsic QoS and does not deal with perceived QoS. It stems from the main objectives of IETF, concerned with the Internet architecture and its development, dependability, and effectiveness.

During the past few years, IETF has devoted a lot of attention to QoS assurance in IP networks. The task force has developed various QoS mechanisms for the Internet, and it has proposed two significant network architectures: IntServ and DiffServ. The basic difference between those two is a level of granularity of independent treatment of flows in the network. It standardized the Resource Reservation Protocol (RSVP) signaling protocol, originally intended for IntServ model implementation and extended later for other purposes. It also developed the notion of IP-QoS architecture as a comprehensive approach to QoS, and proposed several solutions. IETF defines some architecture-independent QoS parameters as well as specific parameters of network components, such as traffic meters, packet markers, droppers, or schedulers, constituting a particular network architecture. The actual work of the IETF also comprises QoS signaling across different network environments, carried out in the Next Steps in Signaling (NSIS) working group[1]. The intention is to re-use, where appropriate, the protocol mechanisms of RSVP, while at the same time simplifying it and applying a more general signaling model.

In the following sections, we will give an overview of the proposed QoS architectures and point out their advantages and disadvantages.

4.1 Integrated Services

The Integrated Services (IntServ) approach aims at providing per flow end-to-end QoS through IP networks [Braden94]. It is based on the assumption that resources must be explicitly managed in order to meet application requirements. This implies that resource reservation and admission control are key building blocks of the service.

In Chapter 1, we defined a flow as a stream of packets with the same source and destination addresses and port numbers. This definition can be extended by adding that all of these packets are with the same QoS demands [Shenker97b]. Nevertheless, the flow is one way and its packets originate from a single user.

The integrated services rely heavily on resource reservation for being able to guarantee end-to-end queuing delays. By sending resource reservation messages across the network, resources for a certain flow are allocated in network elements. The resource reservation is likely to be performed by means of RSVP (Resource reSerVation Protocol) [Braden97]. Resource reservations through the network are maintained by periodic signalling. The period is per default 30 s, but dependent of the equipment vendor it can be also set to a specific value [2].

[1] http://www.ietf.org/html.charters/nsis-charter.html

[2] For example, in Avici RSVP routers this value can be set between one and 300 seconds.

The reference model of the IntServ architecture is depicted in Figure 4.1 and includes four components:

- Resource reservation setup protocol,
- Admission control routine,
- Classifier, and
- Packet scheduler.

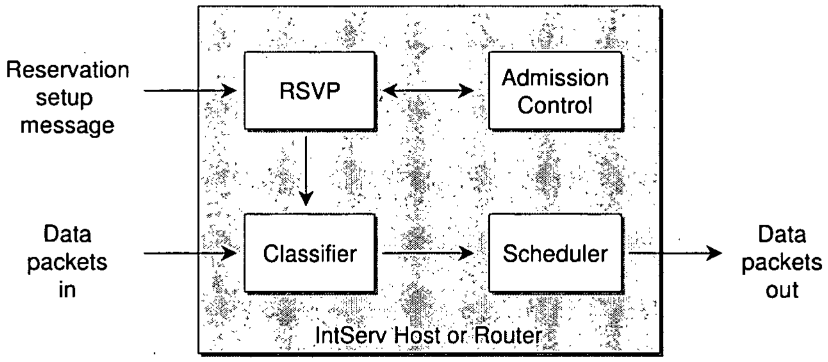

Figure 4.1: Basic components of the integrated services model

Applications requiring a specific service must set up the paths and reserve resources before transmitting their data. Due to this, the end hosts create and maintain their flow-specific state also in routers along the path of a flow. There may be more than one resource reservation protocol to be used in the IntServ model, but it is obvious that multiple choices for reservation protocols will cause confusion. Currently the preferred candidate is the RSVP protocol mentioned above.

Admission control implements the decision algorithm that a router or host uses to determine whether a new flow can be guaranteed the requested QoS without impacting earlier guarantees. Admission control is invoked at each node to make a local accept/reject decision at the time a host requests a service along some path through the network.

When a router receives a packet, the classifier will perform a multifield classification and put the packet in a specific queue based on the classification result. All packets from the same class get the same treatment from the packet scheduler, which manages the forwarding of different packet streams using a set of queues. By this, it should meet the QoS requirements of the flow.

In order to state its resource requirements, the application specifies the desired QoS using a list of parameters that is called flow specification (*flowspec*). The flowspec is carried by the resource reservation setup protocol, passed to admission control to test for acceptability and used to parameterize the packet scheduling mechanism. Filter specifications (*filterspec*) are also carried with the resource reservation setup protocol messages. The filterspec defines the packets that need to be served with this QoS. The filterspec can change without any drawback on the flowspec.

4.1.1 Integrated Services Architecture

In the IntServ architecture (Figure 4.2), each node supports the basic components. There is no difference between the nodes as it will be in the Differentiated Services architecture.

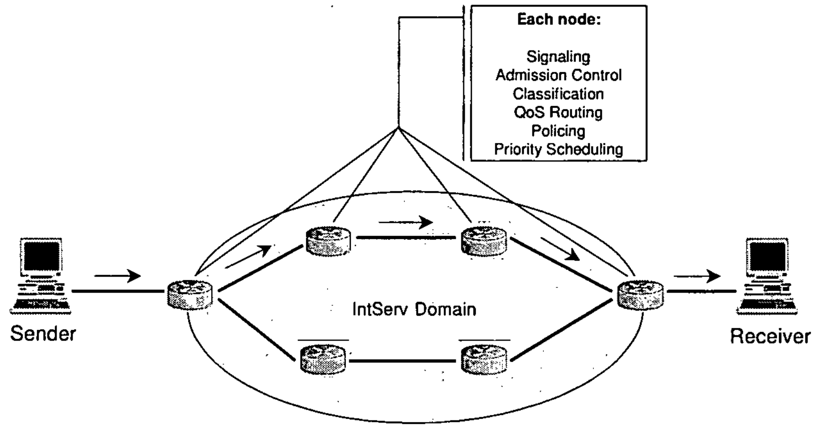

Figure 4.2: Integrated services architecture

Currently three types of services (or traffic classes) have been defined within IntServ :

- The Guaranteed Service [Shenker97a] – is used for providing a bounded queuing delay in IntServ compliant network elements. Furthermore, such network elements will not drop packets with guaranteed service. If all network elements between two communicating entities are IntServ compliant a maximum end-to-end queuing delay will be guaranteed. To achieve this, the client requesting the service must provide the network with both an estimate of which traffic it will generate and the desired service. Non conforming traffic will be dropped. This service is intended for applications that need a firm delay guarantee such as audio and video or real-time multimedia applications.

- The Controlled Load Service [Wroclawski97] – aims at providing the user with service similar to best effort over a lightly loaded network. Admission control is used in the network nodes in order to maintain this impression even under heavy load. To achieve this, the client requesting the service must provide the network with an estimate of the data traffic it will generate. Traffic falling outside the traffic granted by the network may be dropped or delayed. This service is intended to support a broad class of applications which have been developed for use in today's Internet, but are highly sensitive to overload conditions such as adaptive real-time applications.

- Best Effort – is used to provide packets with the best available QoS. There is no guarantee of packet forwarding and packets will be discharged if congestion occurs. Therefore, this class only provides unreliable QoS.

4.1.2 Resource Reservation Protocol

The Resource Reservation Protocol (RSVP) was invented as a signaling protocol for applications to reserve resources [Braden97]. It operates on the concept of flows as defined above. A so-called *flow label* was introduced in the header of the RSVP messages. It provides an association of a flow to RSVP reservations. This flow label is available in the packet header of IPv6 packets. In IPv4, source and destination IP addresses and ports have to be used to identify the flow.

The signaling process in RSVP is illustrated in Figure 4.3. The sender transmits a Path message to the receiver specifying the characteristics of the traffic (the above mentioned flowspec). Every intermediate router along the path forwards the Path message to the next hop determined by the routing protocol. Beside the flowspec, the Path message provides path state information, the IP address of the previous hop node and the sender template (sender IP address, protocol ID, and optional sender UDP/TCP port address).

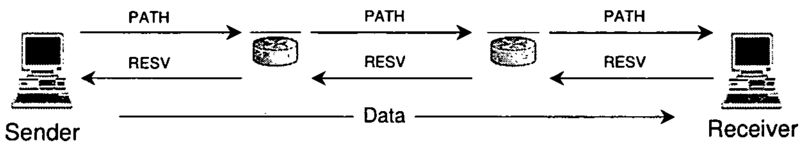

Figure 4.3: RSVP message exchange

Upon receiving a Path message, the receiver responds with a Resv message to request resources for the flow. This message is forwarded hop-by-hop in the sense that each RSVP compliant node forwards the Resv message to the IP address of the next hop provided by the Path message on the way towards the sender. Therefore, the RSVP mechanism is receiver-oriented, in the sense that the receiver has to make the final request on bandwidth reservation. Every intermediate router along the path can reject or accept the request of the Resv message. If the request is rejected, the router will send an error message to the receiver, and the signaling process will terminate. If the request is accepted, link bandwidth, and buffer space are allocated for the flow, and the related flow state information will be stored in the router. Note that RSVP requests the particular QoS, but it is up to the interface queuing mechanism to implement it.

RSVP messages generate *soft-state information* in nodes along the path. This soft-state information has to be refreshed periodically by Path and Resv messages. Otherwise the reservations will expire. Thus, reservation messages will generate much signalling traffic at the network and state information at the routers, which is one of the disadvantages of RSVP. In addition, flow reservation will increase the complexity of scheduling in routers. For these reasons, it is recommended to use RSVP only in networks with a predictable amount of users.

4.1.3 IntServ Disadvantages

The IntServ/RSVP architecture represents a fundamental change to the current Internet architecture, which is founded on the concept that all flow-related state information should be in the end systems. The main problems with the IntServ architecture are:

- State information has to be maintained in each router within the network. The amount of state information and signaling overhead increases proportionally with the number of flows. This places a huge storage and processing overhead on the routers. Therefore, this architecture does not scale well in the Internet core.

- It must be supported by all network elements along the end-to-end path. All routers must have RSVP, admission control, multifield classification, and packet scheduling. This makes IntServ growing to a quite complex undertaking. Nevertheless, nodes not supporting RSVP let RSVP messages pass. This means that from the sender's and receiver's point of view the nodes guarantee something they cannot comply with. Therefore, a connection setup might succeed, but service parameters cannot be guaranteed end-to-end.

- Applications generally do not know how much bandwidth they need. This could lead to frequent modification of reserved bandwidth and additional signalling. By this, the QoS guarantees could change over the course of time.

4.2 Differentiated Services

The basic problem with IntServ is that resources need to be reserved along the path between the communicating entities. This limits the scalability in the sense that a network element may be required to handle a very large amount of reservations. To avoid this problem Differentiated Services (DiffServ) [Blake98] is not based on resource reservation. Instead, it relies on two mechanisms - *per-hop packet handling* and *traffic conditioning*. The basic architecture of DiffServ has been depicted in Figure 4.4.

It consists of a concatenation of DiffServ domains. Each domain may be administrated by different authorities. A DiffServ domain consists of boundary (edge) nodes and interior nodes. Boundary nodes connect DiffServ domains to other DiffServ domains or other domains that are not DiffServ compliant. By providing service information in the IP header, every DiffServ compliant network element treats each packet according to its service requirement. The so-called DS byte is used for carrying the service information [Nichols98]. Traf-

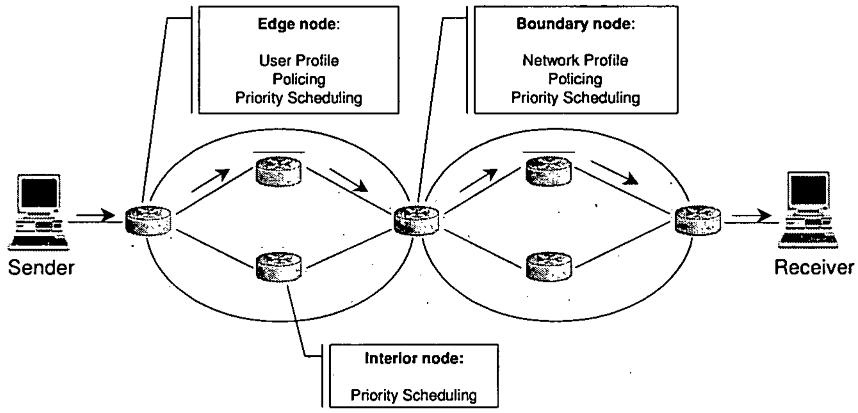

Figure 4.4: Differentiated service architecture

fic conditioning is carried out at the boundaries between networks in order to control the amount of traffic of different service levels that flows from one network to another. The conditioning is performed on aggregate traffic flowing from one network to another according to a Service Level Specification (SLS) and Traffic Conditioning Specification (TCS) between network operators. According to [Grossman02], SLS and TCS are defined as:

- Service Level Specification (SLS) is a set of parameters and their values which together define the service offered to a traffic stream by a DiffServ domain and

- Traffic Conditioning Specification (TCS) is a set of parameters and their values that together specify a set of classifier rules and a traffic profile. A TCS is an integral element of an SLS.

These specifications do not consider pricing, contractual, or other business nature. Even service availability is not addressed by DiffServ. The broader context of an agreement between parties is denoted as Service Level Agreement (SLA) and Traffic Conditioning Agreement (TCA).

As a result of this strategy, DiffServ, where resources are not reserved along the end-to-end connection, provides no absolute guarantees about for instance queuing delays.

4.2.1 Differentiated Services Byte

The Differentiated Services (DS) byte is a common name for the IPv4 Type-of-Service (TOS) byte and for the IPv6 Traffic Class byte. According to [Nichols98] two different fields of the DS byte are defined: A six bits long DSCP and two currently unused (CU) bits (Figure 4.5).

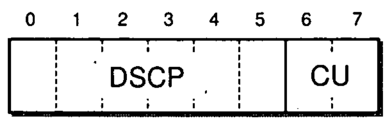

Figure 4.5: Differentiated services byte

In the newest releases, the DiffServ working group of the IETF has been pointed out that this leads to inconsistencies and ambiguities [Grossman02]. In particular, the "Currently Unused" (CU) bits of the DS field have not been assigned to Diffserv, and subsequent to the publication of [Nichols98] (RFC 2474), they were assigned for explicit congestion notification, as defined in [Ramakrishnan01] (RFC 3168). Therefore, the two least significant bits of the IPV4 TOS byte and the IPV6 Traffic Class byte are not used by DiffServ.

The Differentiated Services Code Point (DSCP) field describes the Per-Hop Behavior (PHB) of the packet. The Per-Hop Behavior determines how a packet will be treated by an internal node until it is forwarded towards the next node along the path through the network. The PHB may for instance be implemented by a number of queues with different service priorities. Currently there are three categories of definitions of the PHB [Blake98]:

- Expedited Forwarding (EF),
- Assured Forwarding (AF), and
- Default Behavior (Best Effort).

The default DSCP code for Expedited Forwarding is 101110. Assured Forwarding uses 12 different DSCP codes and the Default Behavior corresponds to Best Effort traffic and is coded by 000000.

Expedited Forwarding

Expedited Forwarding (EF) [Jacobson99] is a PHB that is intended to simulate a virtual leased line. The characteristic of EF is low jitter, low latency, low loss, and assured bandwidth. For the aggregate EF traffic on a node, the achieved departure rate must be equal to or exceed a configurable rate. The EF traffic should receive this rate independent of other traffic transiting the node. To do this, two things needs to be implemented:

- It must be possible to configure the nodes so that the aggregate traffic has a well-defined minimum departure rate and
- The aggregate traffic must be conditioned such that the arrival rate is less than the minimum departure rate.

Traffic that falls outside the traffic profile, i.e., the configured rate, must be dropped for security reasons, since this could signify a denial of service attack or a serious misconfiguration.

Assured Forwarding

Assured Forwarding (AF) [Heinanen99] consists of four classes, where each class has a certain amount of forwarding resources (bandwidth and buffer space) allocated to it. Each class then has three levels of drop preferences. There are two main requirements on the handling of packets in the Assured Forwarding PHB:

- Packets within a class with a drop precedence x must not be forwarded with a smaller probability than those with a drop precedence y, when $x < y$, and
- Packets within a flow may not be re-ordered when they belong to the same AF class.

Traffic conditioning of AF traffic may be done at the edge of a domain. This conditioning can consist of shaping, dropping, and marking of drop precedence and AF class. The traffic conditioning may not cause reordering of packets within a flow. Each DiffServ node must forward packets from different AF classes independently, i.e., AF classes may not be aggregated. Within each AF class all drop precedences must be accepted, but may be aggregated to at least two levels of drop precedence. The default values of DSCP for AF traffic are given in Figure 4.6.

		Class 1	Class 2	Class 3	Class 4
Drop precedence	Low	001010	010010	011010	100010
	Medium	001100	010100	011100	100100
	High	001110	010110	011110	100110

Figure 4.6: Assured forwarding default values of DSCP

4.2.2 Differentiated Services Architecture

There are three key elements in DiffServ networks:

- Internal nodes,
- Edge nodes, and
- Hosts.

Edge nodes are located at the boundary of the network, i.e., towards other networks. Their main task is to perform traffic classification and conditioning of both incoming and outgoing packets according to the SLS/TCS between the network operators. An edge node that only interfaces other DS compliant nodes is also known as a boundary node. In the interior of the network, internal nodes forward individual packets according to their code points in the IP headers. Packets are generated at the hosts, which may or may not be able to set the DS byte according to the service requirements of the application currently used. Ideally, an application running on a host will set the DS byte in the IP header. Network elements along the path through the network will then treat the packet accordingly. However, there may be networks between the communicating peers that use different code points in the DS byte for the same service. In that case, the edge nodes may translate the DS byte. How this translation is done is defined in the SLS/TCS.

4.2.3 Traffic Classification and Conditioning

Traffic classification is performed by a classifier. The purpose of traffic classification is to determine how the traffic is to be handled within the domain. The classification is then used to steer the traffic to a specific part of the traffic conditioner that handles traffic that fulfils the specific parameters of the packet. Traffic conditioning is executed in the DiffServ edge nodes according to a Traffic Conditioning Agreement being part of the Service Level Agreement between network operators. The reason for this mechanism is to control the amount of traffic flowing from one network to another. Therefore, it is applied to aggregate traffic and not to individual flows. Traffic conditioning typically consists of four parts [Blake98]:

- Metering,
- Marking,
- Shaping, and
- Policing.

The aim of traffic conditioning is to enforce rules for the traffic as specified in the TCS. Traffic conditioning is performed at the edge nodes to ensure that the incoming, and sometimes also outgoing, traffic follows the given profile in order to ensure that the aggregate traffic can be handled within the domain.

Figure 4.7 shows the block diagram of a classifier and traffic conditioner. Note that a traffic conditioner may not necessarily contain all four elements. For example, in the case where no traffic profile is in effect, packets may only pass through a classifier and a marker.

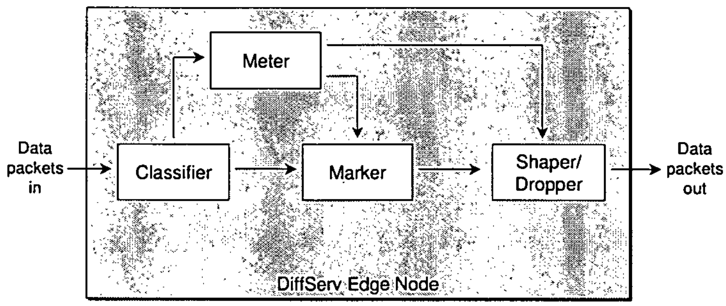

Figure 4.7: Logical view of a packet classifier and traffic conditioner

Classifier

The classifier classifies each packet according to either only the DS byte, so-called BA (Behaviour Aggregate) classifiers, or to a combination of fields, so-called MF (MultiField) classifiers. These fields can be for instance source address, destination address, and protocol ID.

Metering

In order to enforce the Service Level Agreement, process incoming traffic must be metered. The output from the metering is then used in the other parts of the traffic conditioning.

Marker

Three different cases for marking are possible in DiffServ:

- Marking when no PHB has been defined for the packet. In this case, the packet has to be marked with whatever default PHB is given in the Traffic Conditioning Specification (TCS). This could be the default DSCP, i.e., best effort. It could just as well be any other PHB depending on the agreement made with the party sending the information.

- Re-marking for in-profile traffic. This is applicable e.g., when an incoming packet is marked with a PHB that is not supported by the domain. The packet(s) are then re-marked according to what is stated in the TCS.

- Re-marking for out-of-profile traffic. Out-of-profile traffic can be handled in several ways, either by dropping it, re-marking or e.g., just by triggering an additional charge. If re-marking is applied, this shall also be done according to the TCS.

Marking can be carried out both at egress and ingress nodes. This is especially true when non-standard PHBs are set. Normally however, marking will only be done at the ingress node.

Shaping

Shaping is the act of delaying traffic to make it conform to a defined traffic profile. This would normally be achieved by queuing. When queues are getting full, packets will be dropped using e.g., a RED (Random Early Detection) like algorithm.

Policing

When a traffic stream goes outside the limits given by the traffic profile, the packets will be handled according to the TCS. One way of handling the out-of-profile traffic is to discard the packets. This is performed by the policing function. The shaper/dropper in DiffServ can be implemented according to Figure 4.8.

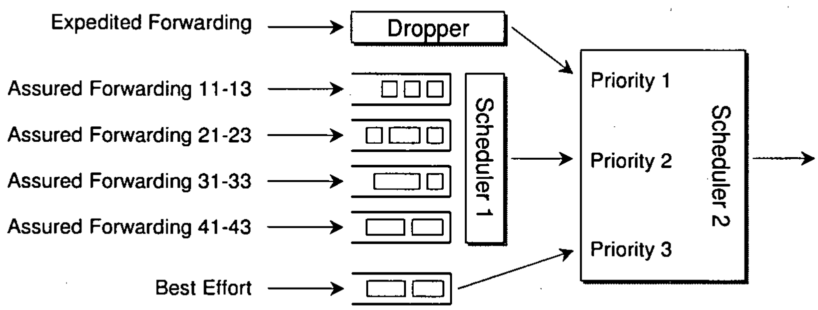

Figure 4.8: Possible implementation of a shaper/dropper for DiffServ

In this scheme, there are two scheduling points. The first is for scheduling between the different AF classes and the second for scheduling between EF, AF, and BE PHB traffic.

The requirements for the EF PHB state that as long as the aggregate EF traffic is within the configured rate, it must be forwarded first. It is also stated that all packets that come outside the configured rate must be dropped. Given these two statements, Scheduler 2 could be implemented using a priority scheduling algorithm. Having BE traffic going directly into this scheduler also ensures that traffic from EF and AF always is served first.

For the AF traffic, four different queues are defined based on AF class. Within these queues Random Early Detection (RED), or a similar concept, should be used to handle congestion. Scheduler 1 should use an algorithm that prevents starvation. The two main algorithms discussed are WFQ (Weighted Fair Queuing) and WRR (Weighted Round Robin).

4.2.4 DiffServ Disadvantages

The DiffServ architecture has been developed in response to some of the drawbacks of the IntServ architecture, but there are also some drawbacks of the DiffServ architecture such as:

- Local congestion in the core cannot be prevented, since DiffServ works without reservation. Therefore, DiffServ does not guarantee absolute bandwidth or delay but only a relative level of QoS.
- Classification of flows is a complex task. The ingress router might have to analyze protocol information of higher layer protocols to determine which DS codepoint to assign. Looking for higher protocol information not only contradicts to the layered model [Carpenter96], but also means a higher effort to decode the (variable-sized) packet header.
- SLSs need to be defined at the domain border to assure that the classes receive the correct QoS in the other domain.

4.3 Multi-Protocol Label Switching

Multi-Protocol Label Switching (MPLS) was originally presented as a way of improving the forwarding speed of routers, but it has capabilities that enable an operator to provide service differentiation[3]. Like IntServ and DiffServ, it is an IETF standard for IP QoS, but its role in QoS is different. IntServ and DiffServ network models do not depend on OSI/ISO layer-2 techniques and define a general QoS architecture for IP networks, which can integrate different transmission techniques in one IP network. MPLS is another networking

[3] It evolved from Cisco's *Tag Switching* and IBM's ARIS (Aggregate Route-based IP-Switching).

technique (like ATM and frame relay) defined in layers 2 and layer 3. The primary goal is to enable layer-2 (link layer) switching speeds at layer 3 (network layer). By this it is possible to enable a specific QoS of connection-oriented traffic inside a connectionless environment (as IP), which is of particular interest for real-time traffic. Currently, its main role is in traffic engineering and virtual private network support.

Some features of MPLS can facilitate QoS assurance. It can extend IntServ and DiffServ capabilities to a wider range of platforms beyond the IP environment. It facilitates offering IP QoS services via FR or ATM networks. Other MPLS features, such as capabilities for load balancing, flow control, explicit routing, and tunnelling, are also important from the QoS perspective.

4.3.1 Basic Principles of MPLS

The basic principle of MPLS is that routers at the edge of the MPLS domain (Label Edge Router, LER) mark all packets with a fixed-length header that acts as shorthand for the information contained in the IP packet header. This header identifies both the route that the packet has to take through the MPLS network and the QoS category of the packet. MPLS packets follow predetermined paths according to traffic engineering and specified QoS levels (Figure 4.9).

The MPLS header is very short and contains a 20-bit label, a 3-bit Class-of-Service (COS) field, a 1-bit label stack indicator, and a 8-bit Time-to-Live (TTL) field. Once within the network, packets can be routed very quickly by an MPLS-capable router, called Label Switched Router (LSR). It examines only the label when forwarding the packet (Label Swapping). This requires significantly less processing than routing based on analysis of an IP packet header. The network protocol can be IP or others. This is why it is called *multi-protocol* label switching.

MPLS needs a protocol to distribute labels to set up Label Switched Paths (LSPs). Whether a generic Label Distribution Protocol (LDP) should be created or RSVP should be extended for this purpose is another issue. MPLS labels can also be piggybacked by routing protocols. A label switched path is similar to an ATM Permanent Virtual Circuit (PVC) and is unidirectional from the sender to the receiver. MPLS routers use the protocol to negotiate

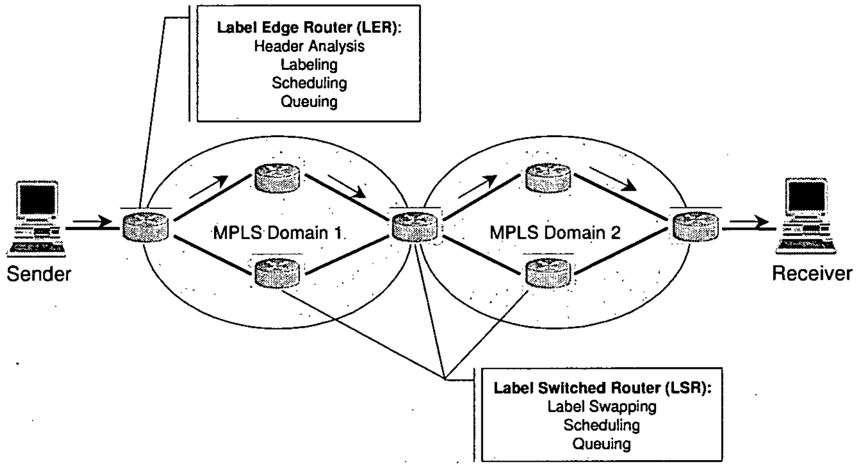

Figure 4.9: MPLS architecture

the semantics of each label, that is, how to handle a packet with a particular label from the peer.

Forwarding Equivalence Class (FEC) is a concept that is introduced in the MPLS standard. An FEC is a class of packets with the same forwarding behavior (routing, forwarding, queuing, etc.). In other words, it is an aggregation of flows with the same service class that can be put into an LSP. By this, only one path is created for each FEC through the MPLS domain. A router negotiates a label for each class with its up and downstream nodes along the path through the domain. This is initiated by the upstream router, which requests a label from its downstream counterpart for a given FEC. *Traffic trunk* is also used as a synonym for FEC.

The setup of the path can be *control-driven* (i.e., triggered by control traffic such as routing updates) or *data-driven* (i.e., triggered by the request of a flow or traffic trunk). The path between two routers can be the same as the layer-3 hop-by-hop route, or the sender router can specify an explicit route for the path. The ability to set up explicit routes is one of the most useful features of MPLS because of the possibility of building Virtual Private Networks (VPNs). A forwarding table indexed by labels is constructed as a result of label distribution. Each forwarding table entry specifies how to process packets carrying the indexing label.

Packets are classified and routed at the ingress routers of an MPLS-capable domain. MPLS headers are then inserted. When a router receives a labelled packet, it will use the label as an index to look up the forwarding table. This is faster than the process of parsing the routing table in search of the longest prefix match as it is done in IP routing [Nilsson97]. The packet is processed as specified by the forwarding table entry. The incoming label is replaced by the outgoing label, and the packet is switched to the next router. Inside an MPLS domain, packet forwarding, classification, and QoS service are determined by the labels and the Class-of-Service fields. This makes core MPLS routers simple. Before a packet leaves an MPLS domain, its MPLS label is removed.

In many respects, MPLS for QoS is similar to the DiffServ approach presented before, although, to reduce the scalability problems, it is usually used as a layer-2 rather than a layer-3 solution. It provides improved granularity of service at the expense of more complex administration. In itself, it cannot provide guaranteed end-to-end QoS configuration on a flow-by-flow basis.

4.4 UMTS QoS Architecture

The third Generation Partnership Programm (3GPP) considers the network services end-to-end when they are from one TE (Terminal Equipment, e.g., laptop or PDA connected to mobile handset) to another TE. It is a layered bearer-service architecture (see Figure 4.10).

Each bearer service on a specific level offers its individual services using services provided by the layers below. To satisfy the end-to-end QoS demanded by the users, several bearer services (possibly located in different networks) have to be set up between the different network elements. These bearer services are TE/MT local bearer service, UMTS bearer service and external bearer service. The external bearer service depends heavily on the characteristics of the external network to which the session is routed. This can be a QoS enabled or best effort IP network. The core network gateway has to offer all required mappings between the UMTS QoS parameters and the specific QoS parameters necessary to offer QoS in each external network connected to it.

The UMTS bearer service provides the various services related with QoS that a UMTS operator may offer. It uses the GPRS PDP (Packet Data Protocol) context procedures to negotiate the QoS parameters that will define the

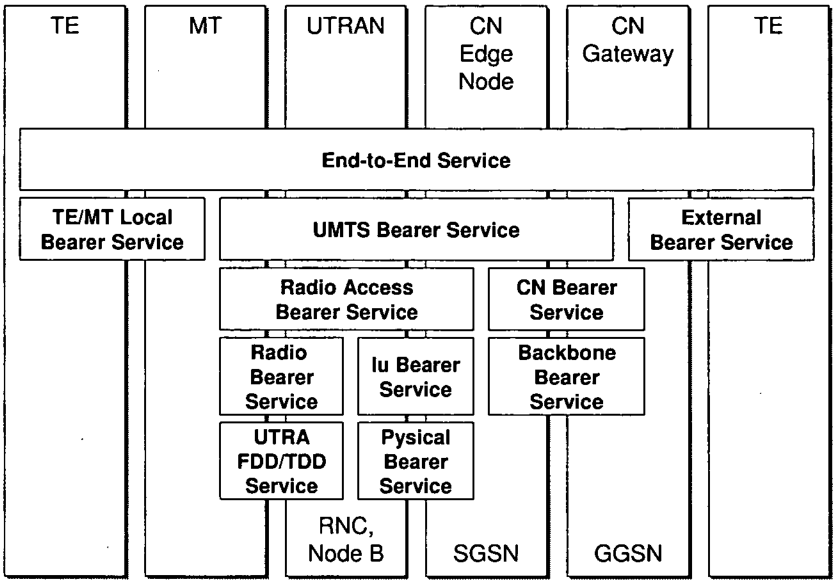

Figure 4.10: UMTS end-to-end QoS architecture

channel between the mobile terminal and the GGSN (Gateway GPRS Support Node). The UMTS bearer service uses the services of both the Radio Access Bearer and the Core Network Bearer. A DiffServ enabled IP Backbone bearer service is used in order to fulfil the QoS requirements of the core network bearer service. The Radio Access Bearer Service provides encrypted transport of signalling and user data between MT and CN Edge Node with the Radio Access QoS adequate to the negotiated UMTS bearer service. This service is based on the characteristics of the radio interface and is maintained for a moving MT.

4.4.1 UMTS QoS Service Classes

When setting up a PDP context the end-user has the choice between four UMTS QoS classes, every class corresponding to a certain set and range of QoS parameters. The four UMTS QoS classes are:

- Conversational,
- Streaming,

- Interactive, and
- Background.

The *conversational class* provides real-time voice applications with the desired service. This class gives a QoS suited for real-time communications and can be described as premium service class. Because of the sensitivity for delays in voice communications the accepted latency is very low. Hence, conversational class marked packets will be served with the highest priorities.

The *streaming class* intends to support streamed applications. These applications demand real-time communications, but accept variable bit rate flow because of built-in buffer functionality. However, the latency between the end-to-end flow must be limited within a defined maximum value.

Interactive classes and *background classes* are characterized by non real-time traffic. These classes aim to support traditional Internet applications (e.g., FTP, telnet, mail, and WWW browsing). The main difference is that interactive class QoS is provided when applications interactively communicate over the network (e.g., WWW browsing) while background classes are meant for non-interactive data transfer (e.g., FTP downloading). Background class packets are less sensitive for time delays than interactive class packets. Therefore, packets with interactive class given QoS have higher priority than background class packets in the network.

4.5 DiffServ in UMTS

There are a couple of differences between QoS as described in DiffServ and as described in UMTS. These apply to traffic conditioning and the number of service categories. A proposal for mapping between the different QoS classes is part of Chapter 5 and regarding the traffic conditioning functions in UMTS and DiffServ, different definitions are used. See Table 4.11 for a comparison of functions.

In UMTS, the GGSN (Gateway GPRS Support Node) will be an edge node towards the Internet Service Provider (ISP). It might also be a boundary node, depending if the service provider supports DiffServ. The SGSN (Serving GPRS Support Node) will be an edge node towards the UMTS radio access network. In some cases, e.g., for a visiting user where the traffic might traverse several

<table>
<tr><th>UMTS</th><th>DiffServ</th><th>Note</th></tr>
<tr><td>Traffic Admission</td><td>-</td><td>No exact correspondence exists in DiffServ. The closest corresponding function for DiffServ would be handling of SLAs.</td></tr>
<tr><td rowspan="2">Translator / Mapper</td><td>Classifier</td><td>The classifier has no directly corresponding function in UMTS. The work done in the classifier maps best to the Translator / Mapper.</td></tr>
<tr><td rowspan="2">Marking</td><td>Marking in UMTS is done in the Translator / Mapper when translating the DiffServ QoS classes to UMTS classes.</td></tr>
<tr><td rowspan="3">Policer</td><td>The UMTS Policer can perform re-marking for traffic that falls outside The SLA.</td></tr>
<tr><td>Policing</td><td>The UMTS Policer can perform dropping of traffic that falls outside the SLA.</td></tr>
<tr><td>Shaping</td><td>The UMTS Policer can delay traffic that falls outside the SLA.</td></tr>
<tr><td>Monitoring</td><td>Metering</td><td>-</td></tr>
</table>

Figure 4.11: UMTS vs. DiffServ traffic classification and conditioning

networks when going from the SGSN in the visited network to the GGSN in the home network, there can also be routers in the core network functioning as edge nodes. There are generally no hosts connected to the core, since the users are connected via the UTRAN.

4.6 IntServ over DiffServ Networks

IntServ and DiffServ compliant networks rely on different mechanism to guarantee end user QoS. That does not necessarily have to be considered as the concepts have to compete. Both mechanisms have their advantages and it is likely to believe that network operators will have to support mechanisms of mapping them. While IntServ are well suited for small networks, DiffServ can fulfil the requirements on QoS in larger networks. Requirements on router support for IntServ and DiffServ can be considered as the basic problem for end-to-end QoS. Routers might not always be aware of the needed mechanism. A framework for RSVP to DiffServ mapping is proposed in [Bernet99]. Another proposed idea at the IETF is to extend the RSVP protocol with a PHB field that carries PHB information. In that case the DSCP settings correspond to the RSVP service classes.

A possible architecture for IntServ over DiffServ domains is based on the IETF draft [Bernet00]. This draft is proposed by the Integrated Services over Specific Link Layer (ISSLL) working group. The ISSLL working group was initially formed to consider how to provide IntServ over specific link technologies, such as a shared Ethernet cable. One of the key ideas that were brought forward from this working group is an approach to provide IntServ QoS by using DiffServ network segments. This solution maintains the IntServ signalling, delay-based admission, and the IntServ service definitions. The edges of the network consist of pure IntServ regions. However, the core of the network is a DiffServ region, and all flows are mapped into one of a few DiffServ classes at the boundary - depending upon the implementation, in either the edge or the border routers of Figure 4.12.

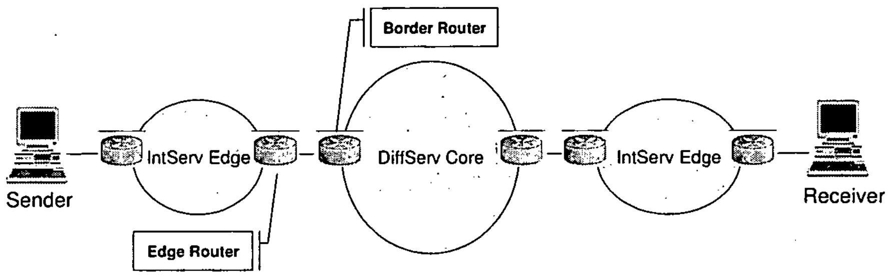

Figure 4.12: IntServ over DiffServ domains architecture

When an RSVP request is received, the edge/boundary node must compare the request with the SLS/TCS that is in place. If the aggregated resources or service class requested are less than the profile permitted by the SLS/TCS entry, then the reservation or service class can be admitted and the RSVP message should be allowed to continue flowing upstream towards the sender. If the resources requested or service class exceed those allowable by the profile in the SLS/TCS entry, the edge node should reject the RSVP request and send a rejection message towards the receiver.

This approach essentially treats the core of the network as a single (logical) IntServ link. This "link" is created by tunnelling (or IPv6 source routing) the data and signalling messages across the DiffServ core. This ensures that routing table updates in the core do not lead to changes in the border/edge routers used by traffic. Traffic conditioning may exist both at the edge of the network and at the DiffServ network boundaries.

The advantage of this solution is that it allows hop-by-hop call admission, and flow-based scheduling at the edges of the network, where low traffic densities make this the most practical way to achieve good quality guarantees. In the core of the network, the scalable solution of DiffServ scheduling can be used, where hard guarantees on QoS can be given on the basis of more probabilistic judgements.

From the above discussion, it can be seen that most QoS architectural solutions may be based around the ISSLL solution, with attention paid to the class definitions.

5 QoS Architecture for LBS in UMTS

The basic simplified UMTS architecture for LBS has been shown in Figure 1.4. Since the UMTS core network interconnects radio access networks, it shall provide QoS mechanisms, which enable interoperability with the QoS architecture for the access part. The focus of this work is the core and backbone part. But as QoS architectures for UMTS should be end-to-end oriented, we are also showing a possible QoS architecture for the access part. Therefore, we will first give a brief overview of the QoS architecture for the access part and afterward discuss the QoS mechanisms in the core of UMTS. Part of this architecture is also end-user QoS and mapping of QoS classes of DiffServ to UMTS.

5.1 QoS in the UMTS Radio Access Part

As shown in Figure 5.1, the access network is implemented as one DiffServ domain. The base station routers acts as edge routers classifying the packets sent in the upstream direction. In addition, they implement shaping and policing to ensure that the traffic injected into the network is not more than that specified through the SLSs between end-users and the network provider. The routers along the path to the Radio Network Controller perform routing and priority scheduling on the aggregated traffic coming from the base stations. Additional policing may be implemented to prevent congestion in the interior nodes. The Radio Network Controller is the border router to the core network. It classifies the packets on the downlink direction and performs policing and shaping to the uplink stream.

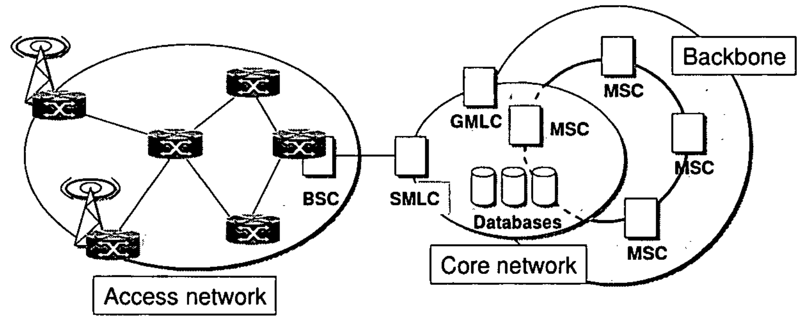

Figure 5.1: QoS architecture in the access network

In addition to best-effort service both Expedited Forwarding and Assured Forwarding PHB are implemented. The proposal includes a resource server to enable on-demand allocation of resources for better bandwidth utilization and congestion control.

5.2 QoS in the UMTS Core Network

The QoS architecture for the access part can be extended to the core network. The core network may be implemented as either one DiffServ domain or be split into different DiffServ domains (e.g., separated GPRS and UMTS domains). Splitting the core into different domains may provide better control of the smaller domains, but adds complexity and extra processing in the interfaces between the domains, since SLSs as well as classification and traffic conditioning functions must be implemented. Implementing the core network as a single DiffServ domain simplifies traffic conditioning and reduces the administrative overhead.

Figure 5.2 illustrates the QoS functions needed to provide DiffServ to the core. The logical entities indicate their individual roles. In reality, a single type of equipment may be used to fulfill multiple roles and may perform multiple functions, e.g., a GGSN to implement Border Gateway (BG) functionality, RNC, or MGW as a Mobile Border Gateway (MBG) to the core network.

The main goal of this architecture is to keep the forwarding path simple while pushing the complexity to the edges of the network to the extent possible.

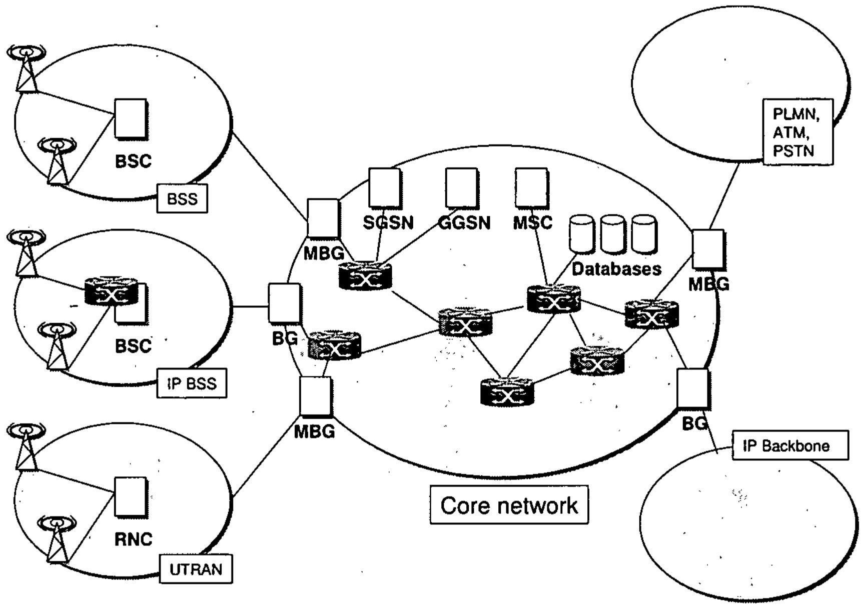

Figure 5.2: QoS architecture in the core network

Furthermore, it provides a service, which is not based on assumptions about the application using it. It therefore employs traffic conditioning that is compatible with both long-term and short-term provisioning. The core network may interface different types of access networks and other types of networks like PSTN, other mobile networks, Internet Service Provider (ISP), corporate networks, or backbones using different transport mechanisms (e.g., ATM, IP). To control the traffic between the networks, Service Level Specifications (SLS) have to be established. Service Level Specifications will be based on bilateral agreements between the operators. Standardizing SLSs will ease the interoperability between the networks but will reduce the flexibility the operators will have, to differentiate their services as well as to implement their own policy and charging schemes. The border nodes must implement traffic conditioning functions (classification, metering, policing, shaping, and scheduling) for traffic entering the core, as described in Section 4.2.

5.3 Service Architecture

The core network implements the Expedited Forwarding Per-Hop Behavior (PHB), the Assured Forwarding PHB, and the Default Behavior. Default Behavior corresponds to best-effort traffic. Services that require guaranteed peak bandwidth service with small queuing and jitter delay are implemented as Expedited Forwarding service (e.g., for voice transfer, real-time video). The Assured Forwarding PHB implements a service that allows bursty behavior, permits longer delays than the Expedited Forwarding service, and provides different levels of differentiation in congestion situations. This service architecture permits traffic flows to make use of any available capacity (e.g., using buffer reserved for Expedited Forwarding service for Assured Forwarding traffic in periods of congestion) without penalty and occasional dropped packets for short congestive periods, which is acceptable by many users, such as non real-time video streams.

5.4 End-User QoS

End-user QoS is assumed to be defined by the application, which sets the DS byte in the IP header. Consequently a mechanism to set the DS byte has to be defined in the mobile station. The mechanism will vary depending on the DiffServ support the applications can provide but also on the support the service provider can provide. However not all applications may take advantage of the DiffServ QoS support. Additional mechanisms are therefore needed to guarantee the users the desired QoS. In the following, some of the possible solutions are given.

5.4.1 Application-Based QoS

In the application-based QoS approach, the application is responsible for demanding the desired service class. Different categories of applications (e.g., streaming video or real-time audio) will request their needed QoS from the network. In that case, the application itself is equipped with a mechanism to forward the request to the underlying IP protocol. However, it must be assumed that not all applications may be able to set the DS byte and other mechanisms are needed. The DS point code set by the end terminal may be reset at the boundary for instance if the current value is not supported in the core network. One possible approach is to introduce an application programming interface (API) in the client that can influence the DS-byte setting. Applications could easily take advantage of the API and forward their QoS demands. Nevertheless, this requires wide support by software vendors and has to be standardized. The application server or the location server can also provide the appropriate value for the DS field to the edge nodes. Since these servers participate in the application signalling, they have the information about the required traffic characteristics. The problem seen, if one left QoS demands to the applications, is that applications may tend to demand the highest possible service class even if a lower class could provide the needed QoS. Therefore, SLS/TCS has to be in place between users and operators as well as between operators.

5.4.2 Multi-field Classification-Based QoS

Multi-field classification-based QoS means that the DS byte is set according to information from more than one field of the TCP/IP header, e.g., port number, IP address. Most applications use a defined port number for communication with other services over the network. The port number defines the endpoint of a logical connection between two services in a network. The protocol header in the UDP and TCP protocol carries the destination port number for each packet while the IP header provides IP address information. This information can be used when edge nodes inspect the protocol headers to provide the needed service class. Care has to be taken for applications that use dynamic port addresses.

5.4.3 Subscription-Based QoS

Subscription-based QoS can be bounded to individual user subscription, application categories, services classes, or positioning information. The application categories might be defined as real-time communication, such as VoIP, streaming class (such as streamed audio or video) and interactive class (such as web browsing or chat applications). These application categories can be divided into even smaller service classes such as gold, silver or bronze classes, and best-effort. When network resources are requested, the network provides the QoS that the user has subscribed to. The subscription information can be stored in the user profile at a policy server. The policy server functionality could be implemented in the HLR. Mapping of subscription-based QoS to packets in the DiffServ domain could then be done at a network node (e.g., Radio Network Controller) for the uplink and either at the Internet Service Provider or at a Mobile Switching Center for the downlink. Since the Radio Network Controller is not aware of HLR data, the core network has to provide this information. QoS mapping will follow the service provider roles and policies and is not part of the standardization. It can be assumed that packet treatment in subscription-based QoS follows the role: the higher the price, the better the service. This does not necessarily mean delay time guarantees but that packets receive proper treatment in the core network.

5.5 QoS Mapping

End-to-end QoS enforcement is rather difficult since packets have to rely on the treatment provided by different QoS domains. The challenge is to provide the desired QoS in the access network, which forms the bottleneck for end-to-end QoS in terms of packet loss and bandwidth. When multimedia and other applications triggered by LBS are an issue, bandwidth demands will raise enormously and the challenge is to satisfy the request. Mapping of IP-based QoS to UMTS QoS classes will result in less differentiation since fewer service classes are defined for UMTS.

5.5.1 Mapping DiffServ to UMTS

The DS-byte setting takes place at the edge of the network, either at the base station or at the MSC when entering the core network. A basic difference between the DiffServ service classes and the UMTS service classes is that in UMTS differentiation within the same class is not foreseen. Consequently fewer service classes will be defined in UMTS/GSM. Table 5.1 shows a possible mapping between UMTS and DiffServ classes.

Table 5.1: Mapping between DiffServ and UMTS service classes

DiffServ	UMTS Service Classes
EF	Conversational
AF1.1 – AF1.3	
AF2.1 – AF2.3	Stream
AF3.1 – AF3.3	Interactive
AF4.1 – AF4.3	
BE	Background

The conversational service in UMTS can be mapped straightforward to the Expedited Forwarding class. Problems can be seen with mapping of the streaming class and interactive class to the AF classes. The AF class consists of twelve service classes while the UMTS QoS only provides two classes. Packets that are served with different priorities in other DiffServ domains may be treated equally in the core network. Background traffic is mapped to the best-effort service.

5.5.2 Mapping between ISP and the Core Network

According to [Blake98] the DS-byte settings can be changed when a packet enters a new DiffServ domain. Relations between DiffServ classes are not defined by the IETF and therefore they will be operator-specific. The intention is to guarantee the same QoS priorities in the core network of UMTS as some other IP networks. However, in the UMTS case, packet treatment in the core network will be based on user subscription data. That does not guarantee bandwidth or throughput but priorities in relation to other user's IP packets. Those QoS priorities are based on the SLS/TCS between operators. Admission control functions need to be implemented in the edge nodes. It will keep track of aggregated flow load and QoS requests to IP addresses according to SLS/TCS information and user subscription data. This information could be provided by the HLR. The DS-byte setting in the transport packets at the downlink needs to be done according to the user subscription data. This is done at the Gateway GPRS Support Node (GGSN). Since the user packet is encapsulated in the transport packet no action has to be taken on user IP-packet QoS settings. On the uplink it is assumed that the user IP packets have a valid DS-byte setting. Otherwise, the edge node (GGSN) may need to set the DS byte before the packet is leaving the core network.

5.5.3 Mapping between the Core Network and the Radio Access Bearer

QoS in UMTS is rather complicated especially when mobility is an issue. There is no guarantee for packet treatment in case of handover. The core network will provide IP service class mapping to UMTS service classes. The GGSN should be equipped with functionality to support mapping towards UTRAN.

The UMTS service classes will be realized with a number of defined Radio Access Bearers (RAB), where each bearer's characteristics correspond to a QoS class. Mapping functions for downlink traffic needs to be implemented in the GGSN. IP packets mapping towards a bearer could then be based on GTP ID, IP address, DS-byte setting, and user subscription data. For uplink traffic, mapping is done in the Radio Network Controller in the sense that packets are encapsulated in a transport IP packet and send through the core. The DS-byte setting needs to be done according to user subscription data. The Radio Network Controller obtains the subscription properties from the core network.

5.5.4 Mapping between UE and CN

The user equipment has to request access to the core network from the SGSN before packets can be transmitted. In that case, negotiations need to be done between the access and the core. PDP-context (Packet Data Protocol) information, carrying QoS properties are sent over the network towards the SGSN. If the core network accepts the request and radio resources are available, they will respond to the demand. Then, a bearer is provided between the client and the SGSN. A bearer consists of an RLC (Radio Link Control) link between the Radio Network Controller and the user equipment and a tunnel between the Radio Network Controller and the SGSN. User equipment can handle several bearers simultaneously. With respect to bearer characteristics, packets mapped to the same tunnel will be served with the same priorities. Therefore, packet flows must be mapped to the tunnel that corresponds to the desired priorities.

6 Scheduling

6.1 Introduction

Scheduling algorithm are seen as a small but critical component in the design of router and with an important role in the QoS provisions. Actually, every packet-switched telecommunication network allows users to share bandwidth. Sharing automatically introduce the problem of contention for the shared resource. Given a set of resource request in the service queue, a router (or generally a server) uses a scheduling discipline to decide which request to serve next. Scheduling disciplines are very important because they ar the key to fairly sharing the network bandwidth and to providing performance-critical applications with performance guarantees. This applications can be voice or video-conferences but also the broad range of location-based applications.

Data traffic, as well as some real-time applications like compressed video, may exhibit a much higher peak transmission rate than their average rate. Thus, the peak rate allocation in circuit-switching results in poor utilization of the link bandwidth. Therefore, packet-switching or store-and-forward network are the option as the alternative so that applications may transmit bursts of data, with a peak-rate much higher than their average rate, though the network resources are allocated with the average transmission rate of the applications. Buffering is required in the intermediate nodes of the network (switches or routers) to absorb traffic bursts and avoid packet loss. The above approach is based on the key principle of *statistical multiplexing* [Bertsekas92].

In the context of statistical multiplexing, the packet-switching network can be operated in an efficient way, but the possible congestion may degrade their performance in either short or long time intervals. Network users may become synchronized, transmitting their maximum bursts at the same time interval and the network may need very large buffers to absorb these bursts in each node. On the other hand, by introducing large buffers, the queues would grow without bound, and the end-to-end delay would increase. Moreover, when packet lifetime is finite, the packets leaving the router would have timed out already and been retransmitted by the transport protocols. Users may also misbehave, trying to use more than a fair share of the network resources or more than what they have reserved. Thus in order to balance the network performance and efficiency, the mechanism of traffic management must be introduced. Specially, packets from different connections will interact with each other at each switch; without proper control, these interactions may adversely affect the network performance experienced by users.

Traffic management is the set of policies and mechanisms that allow a network to efficiently satisfy a diverse range of service requests. The two fundamental aspects of traffic management, diversity in user requirements and efficiency in satisfying them, counteract, creating a tension that has led to a rich set of mechanisms, whereof scheduling disciplines are one of them.

A scheduling discipline actually has two orthogonal components. First, it decides the order in which requests are serviced (service discipline). Second, it manages the service queue of requests awaiting service (buffer management). If more packets are coming to the input of a router than the output capacity is capable to transmit, some packets must wait in the queue, taking up storage space (buffer capacity). If the buffer is limited (what practically is always the case) the router must eventually drop some packets. Which ones should the router drop? In the same way as a scheduling discipline allocates different delays to different users by its choice of service order, it allocates different loss rates to different users by its choice of which packets to drop.

By this we come to an important law:

$$\sum_{i=0}^{n-1} \lambda_i \cdot d_i = constant \tag{6.1}$$

This law is called the *conservation law* and is valid for all work-conserving schedulers[1]. The sum of the scalar product of the mean arrival rate λ_i and

[1] A scheduler is work-conserving if the server is never idle when a packet is buffered in the system.

the mean waiting time d_i of each flow, over all active flows, is always constant. With other words, the conservation law says that some flows can receive lower delay only at the expense of longer delay for other flows. This is particularly evident in priority queuing mechanisms like Weighted Round Robin (WRR) e.g., Class Base Queuing (CBQ). One of the focuses of this work is also by studying fair queuing schedulers, like Weighted Fair Queuing (WFQ), where the delay is fairly distributed among all competing (active) flows. According to [Keshav97] a scheduling discipline must satisfy four, sometimes contradictory, requirements:

- Ease of implementation (for both guaranteed-service and best-effort connections)
- Fairness and protection (for best-effort connections)
- Performance bounds (for guaranteed-service connections)
- Ease of efficiency of admission control (for guaranteed-service connections)

Each scheduling discipline makes a different trade-off among these requirements. Depending on the situation, some of these requirements are more important than others.

The range of choices available in commercial switches is very limited. Most schedulers are First-In-First-Out (FIFO), with a single level of priority (this is true of the vast majority of routers in the Internet). Some switches and routers allow multiple levels of priority and some provide more interesting alternatives, such as Weighted Round Robin (WRR) and Weighted Fair Queuing (WFQ). Yet, manufacturers are realize the need for good scheduling disciplines, and this has resulted in excellent research in this area.

6.2 Classification of Schedulers

In general, schedulers can be characterized as:

- Work-conserving,
- Non work-conserving.

A scheduler is work-conserving if the server is never idle when a packet is buffered in the system. A non wok-conserving server may remain idle even if there are available packets to transmit. Even if non work-conserving disciplines occurs as somehow strange because not using idle resources, the reason for doing it is to make the traffic arriving at downstream switches more predictable, thus reducing both the buffer size necessary at output queues, and the delay jitter experienced by a connection [Keshav97].

Secondly, a server may postpone the transmission of a packet when it expects a higher-priority packet to arrive soon and non work-conserving algorithms are also used to control delay jitter by delaying packets that arrive early [Verma91]. In this way, non-work-conserving schedulers have exhibited some unique advantages in providing guaranteed performance services but work-conserving schedulers are more efficient in providing best-effort service. Work-conserving servers always have lower average delays than non work-conserving servers. Examples of work-conserving schedulers are given in Table 6.1.

Scheduler	Name	Reference
GPS	General Processor Sharing	Parekh92b
WFQ	Weighted Fair Queuing	Demers89
VC	Virtual Clock	Zhang89
WF^2Q	Worst-case Fair Weighted Fair Queuing	Bennett96
Delay-EDD	Delay Earliest Due Date	Ferrari90
WRR	Weighted Round Robin	Katevenis91
DRR	Deficit Round Robin	Shreedhar95

Table 6.1: Examples of work-conserving schedulers

On the other hand, Hierarchical Round Robin (HRR) [Kalmanek90], Stop-and-Go Queuing [Golestani91], and Jitter Earliest Due Date [Verma91] are non work-conserving schedulers.

Another classification of schedulers is based on their internal structure [Zhang91]. According to this classification, there are two main architectures:

- Time-based,
- Frame-based.

In a time-based scheduler, there is a global variable - usually referred to as the virtual time - associated with each outgoing link of the switch. Each time, a packet arrives or departs the system, this variable is updated. A timestamp, computed as a function of this variable, is associated with each packet. Packets are sorted based on their timestamps, and are transmitted in that order. Thereby, this group of schedulers is also known as sorted-based.

Examples of time-based schedulers are given in Table 6.2

Scheduler	Name	Reference
VC	Virtual Clock	Zhang89
WFQ	Weighted Fair Queuing	Demers89
Delay-EDD	Delay Earliest Due Date	Ferrari90

Table 6.2: Examples of time-based schedulers

In a frame-based scheduler, time is split into frames of fixed or variable length. Reservations of sessions are made in terms of the maximum amount of traffic the session is allowed to transmit during a frame period. Hierarchical Round Robin (HRR) and Class Based Queuing (CBQ) are frame-based schedulers that use a constant frame size. As a result, the server may remain idle if sessions transmit less traffic than their reservations over the duration of a frame.

	Scheduler	Name	Reference
Fixed length	HRR	Hierarchical Round Robin	Kalmanek90
	CBQ	Class Base Queuing	Floyd95
Variable length	WRR	Weighted Round Robin	Katevenis91
	DRR	Deficit Round Robin	Shreedhar95

Table 6.3: Examples of frame-based schedulers

In contrast, Weighted Round Robin (WRR) and Deficit Round Robin (DRR) schedulers allow the frame size to vary within a maximum. Thus, if the traffic from a session is less than its reservation, a new frame can be started early. Therefore, both of these schedulers are work-conserving. The listed frame-based schedulers are presented in Table 6.3.

Time-based scheduling schemes are usually more complex than frame-based schemes. However, time-based scheduling schemes are generally more fair and provide lower latency than frame-based schemes. In Table 6.4 a classification of all presented schedulers is given.

6.3 Basic Scheduling Disciplines

In this part the basic principles of first-in first-out, processor sharing, general processor sharing, priority queuing, fair queuing and the earliest due date

Scheduler	Work conserving	Non work-conserving	Time based	Frame based
FIFO	+		+	
PS	+		+	
GPS	+		+	
PQ	+		+	
FQ	+		+	
EDD	+		+	
WRR	+			+
DRR	+			+
HRR		+		+
WFQ	+		+	
CBQ	+			+
SCFQ	+		+	
STFQ	+		+	
WF^2Q	+		+	
SFQ	+		+	
VC	+		+	
CSFQ	+		+	
Delay-EDD	+		+	
Jitter-EDD		+	+	
EDD/FB	+		+	
SC/EDD	+		+	

Table 6.4: Classification of schedulers

scheduling are presented. There are the basics for scheduling disciplines that follows in the next section.

6.3.1 First-In First-Out

First-in First-out (FIFO) or First-Come First-Serve (FCFS) is one of the simplest scheduling policies. Its operation is that packets are served in the order in which they are received. In Figure 6.1, all packets from the active queues are sorted into the server queue and the order depends only on their arrival time.

FIFO is quite simple to implement. In particular, insertion and deletion from

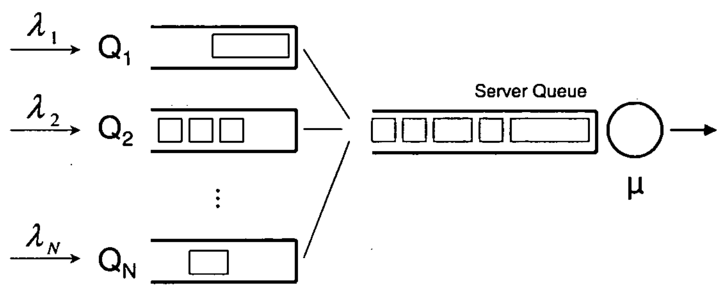

Figure 6.1: First-in First-out

the queue of the waiting packets are constant time operations and do not require any per-connection state to be maintained by the scheduler, so it is one of the most commonly implemented policies. However, the FIFO scheduling does not readily provide delay or throughput guarantees because it does not provide any isolation between individual sessions. One way to provide a delay bound is to limit the buffer size of the FIFO queue. In this way, once a packet is queued up for transmission it is guaranteed to be sent out in the time it takes to serve a full queue of packets, but on the other hand, packets have to be dropped if the limited buffer is full when packets arrive. Though FIFO has an easy operation, it does not explicitly provide any mechanisms for fair sharing of link resources, but with the help of some buffer management schemes it is possible to control the sharing of bandwidth [Pan00, Anjum99, Floyd93, Lin97, Ott99].

FIFO Benefits and Limitations

FIFO queuing offers the following benefits:

- In case of software-based routers, FIFO queuing places an extremely low computational load on the system.
- The behavior of a FIFO queue is very predictable – packets are not reordered and the maximum delay is determined by the maximum depth of the FIFO queue.
- As long as the server queue depth remains short, FIFO queuing provides simple contention resolution for network resources without adding significantly to the queuing delay experienced at each hop.

FIFO queuing also poses the following limitations:

- A single FIFO queue does not permit to serve one class of traffic differently from other classes of traffic.
- A single FIFO queue impacts all flows equally, because the mean queuing delay for all flows increases as congestion increases. As a result, FIFO queuing can result in increased delay, jitter, and loss for real-time applications traversing a FIFO queue.
- During periods of congestion, FIFO queuing benefits UDP flows over TCP flows. When experiencing packet loss due to congestion, TCP-based applications reduce their transmission rate, but UDP-based applications remain oblivious to packet loss and continue transmitting packets at their usual rate. Because TCP-based applications slow their transmission rate to adapt to changing network conditions, FIFO queuing can result in increased delay, jitter, and a reduction in the amount of output bandwidth consumed by TCP applications traversing the queue.
- A bursty flow can consume the entire buffer space of a FIFO queue, and that causes all other flows to be denied service until after the burst is serviced. This can result in increased delay, jitter, and loss for the other well-behaved TCP and UDP flows traversing the queue.

FIFO Implementations and Applications

Generally, FIFO queuing is supported on an output port when no other queue scheduling discipline is configured. In some cases, router vendors implement two queues on an output port when no other queue scheduling discipline is configured: a high-priority queue that is dedicated to scheduling network control traffic and a FIFO queue that schedules all other types of traffic.

6.3.2 Processor Sharing

Processor sharing (PS) or Head-of-Line Processor Sharing (HOL-PS) was originally proposed as an idealization of time-slicing in computer systems [Kleinrock76], where the server cycles through the active jobs, giving each a small time quantum of service, and then preempting the job to work on the next. In data network, the counterpart of time-slicing is not feasible. A job corresponds to a transmission of a packet, which is constructed at the packet's source. It would violate the communications protocol if a node in the interior of the network were to cut short a packet transmission - preempting the job's service - to switch to the partial transmission of another packet. On the other hand, a real server cannot service N sessions simultaneously. In this sense, processor sharing is not implementable.

However, it has been found that mechanisms like PS, if appropriately emulated and combined with sensible buffer and rate control mechanisms, provide network-wide small delay for short communications, fair bandwidth allocation, protection against streams that would otherwise swamp the network, and stable overall network performance, especially if traffic is heavy [Demers89, Zhang89, Hahne91].

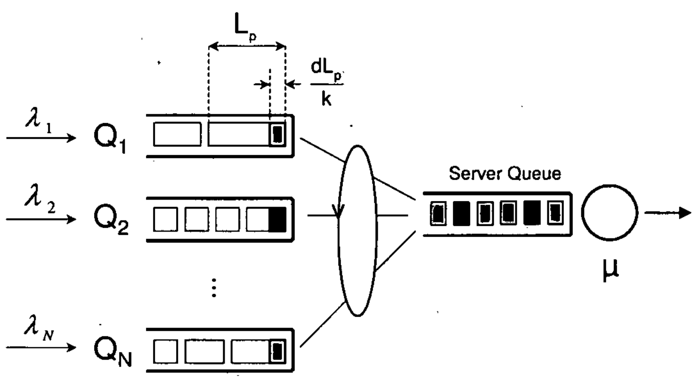

Figure 6.2: Processor sharing

To define PS, we are considering the model depicted in Figure 6.2. There is a single server with a server rate μ, and N job arrival streams, each feeding a different FIFO queue. In the PS discipline, in any interval of time when exactly $k > 0$ queues ar not empty, the server serves the job at the head of each simultaneously (round robin) at rate proportional to $1/k$. In every cycle an infinitesimal small amount of the head-of-line packet, $\frac{L_p}{dL}$, is processed.

Thereby the service rate of session i is defined by:

$$r_i = \mu \cdot \frac{\frac{L_p}{dL}}{\sum activeQueues} \tag{6.2}$$

whereby the number of backlogged queues ($\sum activeQueues$) at time t is often described as $B_{PS}(t)$.

Considering the fairness of any emulation of PS, Demers et al. in [Demers89] tie fairness to throughput and appeal to the fact that PS achieves max-min fair throughput. Formally, we define the *max-min fair share allocation* to be as follows:

- Resources are allocated in order of increasing demand
- No sources gets a resource share larger than its demand
- Sources with unsatisfied demands get an equal share of the resources

In the example depicted in Figure 6.2 we are supposing that the server works at rate 1, and that stream Q_1 receives jobs at rate λ_1 in bits per unit time. Let

$$\phi_1 \geq 0$$

be the achieved throughput or output rate of stream Q_1 under PS. If

$$\lambda = \textstyle\sum_{i=1}^{N} \lambda_i < 1,$$

then simply

$$\phi_i = \lambda_i,$$

for $i = 1, ..., N$. If

$$\lambda \geq 1,$$

then

$$\phi_i = \min\{\lambda_i, \phi_{fair}\},$$

where ϕ_{fair} is determined by the constraint

$$\sum_{i=1}^{N} \phi_i = 1.$$

A stream i whose arrival rate λ_i lies below the threshold ϕ_{fair}, achieves throughput λ_i; otherwise it achieves throughput ϕ_{fair}.

A second requirement for a proper emulation of PS is that it handles arbitrary distributions of job sizes. Many data networks allow packets of variable size. In particular, the Internet Protocol (IP) handles a range of packet lengths (between 21 and 1500 Bytes). Variable length packets rule out emulating PS by serving jobs round robin, that is cycling through the nonempty queues and in one cycle serving just the job at the head of each. If the streams ar not homogeneous then round robin scheduling is not fair as can be shown with simulation results (Figure 6.3).

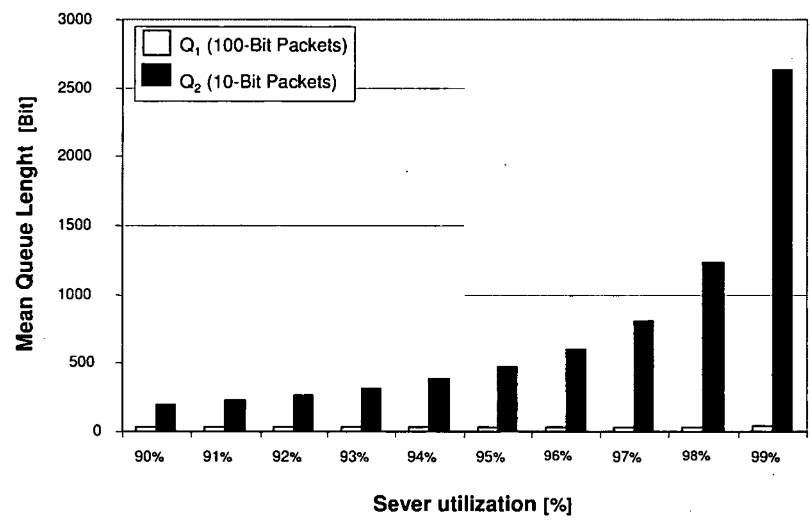

Figure 6.3: Simulation results of round robin scheduling

In the simulation two arrival streams, Q_1 and Q_2 are considered, both with 10 Mbit/s, but with packets length of 100 and 10 bits, respectively. The resulting figure shows the buffer size for Q_1 and Q_2. The buffer requirements for the stream Q_2 are much higher than for Q_1, because Q_1 is getting 10-time more service then Q_2 even both have the same arrival rate of 10 Mbit/s. This is even more evident when the utilization of the server is higher. In [Demers89] a new queuing discipline was introduced, termed Fair Queuing, which attempts to emulate the PS discipline, handles variable job (or packet) sizes, and never preempts a job in service. Simulation results in Figure 6.4 emphasize the fairness of fair queuing with the same streams Q_1 and Q_2 as in the previous example.

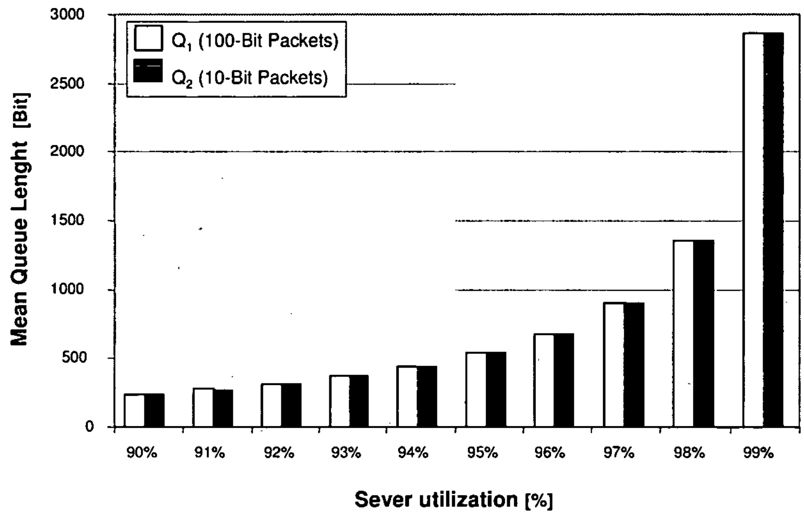

Figure 6.4: Simulation results of fair queuing scheduling

The disadvantage of fair queuing algorithms is there complexity, which is elaborated for Weighted Fair Queuing in Chapter 7. Newest research results in [Wangdong03] show also that round robin algorithms with enough additional bandwidth can match fair queuing algorithms in delay performance.

6.3.3 Generalized Processor Sharing

General Processor Sharing (GPS) [Parekh92b] is a general form of the processor sharing (PS) service discipline. With processor sharing, there is a separate FIFO queue for each connection sharing the same link. During any time interval when there are exactly N non-empty queues, the server serves the N packets at the head of the queues simultaneously, each at a rate of $1/N^{th}$ of the link speed. While a processor sharing server serves all non-empty queues at the same rate, the generalized processor sharing server allows different connections to have different service shares, W_i (Figure 6.5).

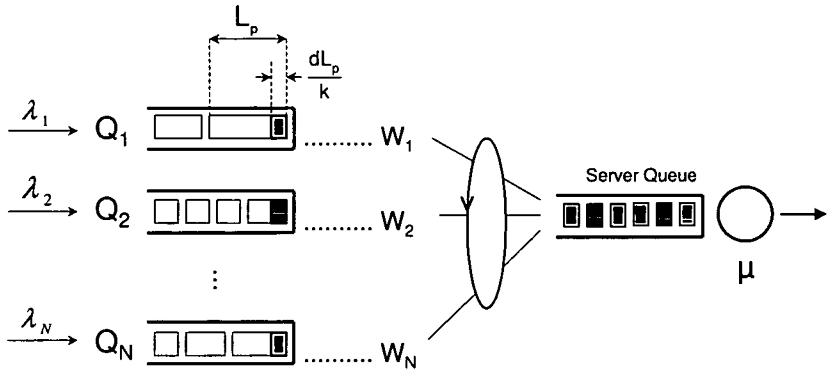

Figure 6.5: General processor sharing

Let again ϕ_i be the amount of the achieved throughput or output rate of stream Q_i and $\phi_i(t_1, t_2)$ the traffic served in the interval $[t_1, t_2]$. By this a generalized processor sharing server is defined as one for which

$$\frac{\phi_i(t_1, t_2)}{\phi_j(t_1, t_2} \geq \frac{W_i}{W_j} \qquad j = 1, 2, ..., N \tag{6.3}$$

holds for any session i that is backlogged throughout the interval $[t_1, t_2]$ [Parekh92a]. From the definition, it immediately follows that if $B_{GPS}(t)$, the set of backlogged sessions in GPS at time t, remains unchanged during any time interval $[t_1, t_2]$, the service rate of session i during the interval will be exactly

$$r_i(t_1, t_2) = \frac{\frac{L_p}{dL}}{B_{GPS}(t)} \cdot \frac{W_i}{\sum_{j \epsilon B_{GPS}(t)} W_j} \cdot \mu \tag{6.4}$$

where $\frac{L_p}{dL}$ is an infinitesimal small amount of the head of line packet. Since $B_{GPS}(t)$ is a subset of all sessions at the server, it follows that that

$$r_i(t_1, t_2) \geq r_i \tag{6.5}$$

holds where

$$r_i = \frac{W_i}{\sum_{j=1}^{N} W_j} \cdot \mu. \tag{6.6}$$

Therefore, session i is guaranteed a minimum service rate of r_i during any interval when it is backlogged. Let the time interval length go to zero, we get the instantaneous service rate of the session, $r_i(t)$.

GPS serves each backlogged session with a minimum rate equal to its reserved rate at each instant; in addition, the excess bandwidth available from sessions not using their reservation is distributed among all the backlogged connections at each instant in proportion to their individual reservation. By doing so, the concept of max-min fair share should be extended to include such weights by defining the max-min weighted fair share allocation as follows [Keshav97]:

- Resources are allocated in order of increasing demand, normalized by the weight,
- No source gets a resource share larger than its demand, and
- Sources with unsatisfied demands get resource shares in proportion to their weights.

A separate thread of studying GPS related disciplines is in the context of providing guaranteed bounded delay services in packet-switched networks. Parekh in [Parekh92a] demonstrate that, by employing GPS servers at switches, end-to-end delay bound can be guaranteed to a session provided its traffic is token bucket constrained at the source. In 1989, he also proposed a packet approximation algorithm for GPS which he called Packet-by-Packet Generalized Processor Sharing (PGPS). PGPS is identical with the independently invented well-known Weighted Fair Queuing (WFQ) [Demers89].

While GPS is ideal in that it exactly achieves a max-min fair allocation, it is like Processor Sharing, unimplementable. Once more, a real server cannot service all N backlogged sessions simultaneously. A number of implementable scheduling disciplines, like Weighted Round Robin (WRR), Deficit Round Robin (DRR), Weighted Fair Queuing (WFQ) and variants of WFQ, are all based on approximating GPS and explained in Section 6.4

6.3.4 Priority Queuing

Priority queuing (PQ) is the basis for a class of queue scheduling algorithms that are designed to provide a relatively simple method of supporting differentiated service classes. In classic priority queuing, packets are first classified by the system and then placed into different priority queues (Figure 6.6). Packets are scheduled from the head of a given queue only if all queues of higher priority are empty. Within each of the priority queues, packets are scheduled in FIFO order.

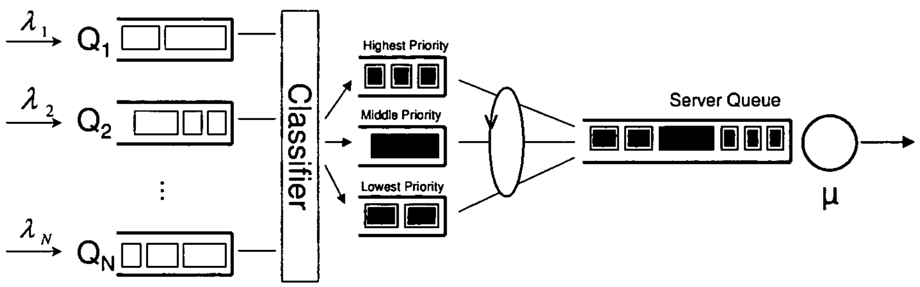

Figure 6.6: Priority queuing

Priority Queuing Benefits and Limitations

PQ offers a couple of benefits:

- For software-based routers, PQ places a relatively low computational load on the system when compared with more elaborate queuing disciplines.
- PQ allows routers to organize buffered packets, and then service one class of traffic differently from other classes of traffic. For example, one can set priorities so that real-time applications, such as interactive voice and video, get priority over applications that do not operate in real time. It is also to the best advantage for signaling messages if they are transmitted over the same network as data.

However PQ also results in several limitations:

- If the amount of high-priority traffic is not policed or conditioned at the edges of the network, lower-priority traffic may experience excessive delay as it waits for unbounded higher-priority traffic to be serviced.
- If the volume of higher-priority traffic becomes excessive, lower-priority traffic can be dropped as the buffer space allocated to low-priority queues

starts to overflow. If this occurs, it is possible that the combination of packet dropping, increased latency, and packet retransmission by host systems can ultimately lead to complete resource starvation for lower-priority traffic. Strict PQ can create a network environment where a reduction in the quality-of-service delivered to the lower-priority service is delayed until the entire network is devoted to processing only the highest-priority service class.

- A misbehaving high-priority flow can add significantly to the amount of delay and jitter experienced by other high-priority flows sharing the same queue.

PQ is not a solution to overcome the limitation of FIFO queuing where UDP flows are favored over TCP flows during periods of congestion. If attempting to use PQ to place TCP flows into a higher-priority queue than UDP flows, TCP window management and flow control mechanisms will attempt to consume all of the available bandwidth on the output port, thus starving lower-priority UDP flows.

Priority Queuing Implementations and Applications

Typically, router vendors allow PQ to be configured to operate in one of two modes:

- Strict priority queuing,
- Rate-controlled priority queuing.

Strict PQ ensures that packets in a high-priority queue are always scheduled before packets in lower-priority queues. Of course, the challenge with this approach is that an excessive amount of high-priority traffic can cause bandwidth starvation for lower priority service classes. However, some carriers may actually want their networks to support this type of behavior. For example, assuming a regulatory agency requires that, in order to carry voice over IP (VoIP) traffic, a service provider must agree not to drop VoIP traffic in order to guarantee a uniform quality-of-service, no matter how much congestion the network might experience. The congestion could result from imprecise admission control leading to an excessive amount of VoIP traffic or, possibly, a network failure.

This behavior can be supported by using strict PQ without a bandwidth limitation, placing VoIP traffic in the highest-priority queue, and allowing the VoIP queue to consume bandwidth that would normally be allocated to the lower-priority queues, if necessary. A provider might be willing to support this type of behavior if the penalties imposed by the regulatory agency exceed the rebates it is required to provide other subscribers for diminished service.

Rate-controlled PQ allows packets in a high-priority queue to be scheduled before packets in lower-priority queues only if the amount of traffic in the high-priority queue stays below a user-configured threshold. For example, we assume a high-priority queue that has been rate-limited to 20% of the output port bandwidth. As long as the high-priority queue consumes less than 20% of the output port bandwidth, packets from this queue are scheduled ahead of packets from lower-priority queues. However, if the high-priority queue consumes more than 20% of the output port bandwidth, packets from lower-priority queues can be scheduled ahead of packets from the high-priority queue. When this occurs, there are no standards, so each vendor determines how its implementation schedules lower-priority packets ahead of high-priority packets.

There are two primary applications for PQ at the edges and in the core of the network:

- PQ can enhance network stability during periods of congestion by allowing to assign routing-protocols and other types of network-control traffic to the highest-priority queue.
- PQ supports the delivery of a high-throughput, low-delay, low-jitter, and low-loss service class. This capability allows to deliver real-time applications, such as interactive voice or video, or to support TDM circuit emulation by giving priority to these services before all other applications.

However, support for these types of services requires to effectively condition traffic at the edges of the network to prevent high-priority queues from becoming oversubscribed. The real challenge lies in the fact that it is much easier to condition traffic and allocate bandwidth to a queue for certain applications than for other applications. For example, it is much easier to provision resources for a well-defined application, such as VoIP, where the packet size, traffic volume, and traffic behavior is known, than it is to provision resources for other types of applications, such as interactive video, where there are just too many variables. It is the presence of these unknowns that makes it extremely difficult to configure traffic conditioning thresholds, maximum queue depths, and bandwidth limits for high-priority queues.

6.3.5 Fair Queuing

Fair queuing (FQ) was proposed by John Nagle in 1987 [Nagle87]. FQ is the foundation for a class of queue scheduling disciplines that are designed to ensure that each flow has fair access to network resources and to prevent a bursty flow from consuming more than its fair share of output port bandwidth. In FQ, packets are first classified into flows by the system and then assigned to a queue that is specifically dedicated to that flow (Figure 6.7). Queues are then serviced one packet at a time in round-robin order. Empty queues are skipped. FQ is also referred to as per-flow or flow-based queuing.

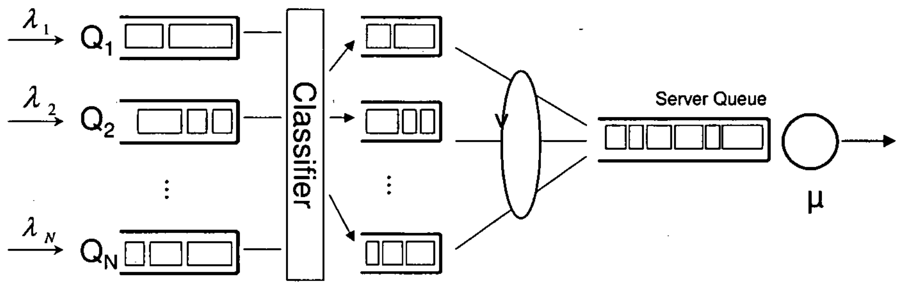

Figure 6.7: Fair queuing

FQ Benefits and Limitations

The primary benefit of FQ is that an extremely bursty or misbehaving flow does not degrade the quality-of-service delivered to other flows, because each flow is isolated into its own queue. If a flow attempts to consume more than its fair share of bandwidth, then only its queue is affected, so there is no impact on the performance of the other queues on the shared output port.

FQ also involves several limitations:

- Vendor implementations of FQ are implemented in software, not hardware. This limits the application of FQ to low-speed interfaces at the edges of the network.

- The objective of FQ is to allocate the same amount of bandwidth to each flow over time. FQ is not designed to support a number of flows with different bandwidth requirements.

- FQ provides equal amounts of bandwidth to each flow only if all of the packets in all of the queues are the same size. Flows containing mostly large packets get a larger share of output port bandwidth than flows containing predominantly small packets.

- FQ is sensitive to the order of packet arrivals. If a packet arrives in an empty queue immediately after the queue is visited by the round-robin scheduler, the packet has to wait in the queue until all of the other queues have been serviced before it can be transmitted.

- FQ does not provide a mechanism that allows to easily support real-time services.

- FQ assumes that one can easily classify network traffic into well-defined flows. In an IP network, this is not as easy as it might first appear. Flows can be classified based on a packet's source address, but then each workstation is given the same amount of network resources as a server or mainframe. Attempting to classify flows based on the TCP connection, one have to look deeper into the packet header, and still have to deal with other issues resulting from encryption, fragmentation, and UDP flows. Finally, considering classifying flows based on source/destination address pairs gives an advantage to servers that have many different sessions, but still provides more than a fair share of network resources to multitasking workstations.

- Depending on the specific mechanism used to classify packets into flows, FQ generally cannot be configured on core routers, because a core router would be required to support thousands or tens of thousands of discrete queues on each port. This increases complexity and management overhead, which adversely impacts the scalability of FQ in large packet networks.

FQ Implementations and Applications

FQ is typically applied at the edges of the network, where subscribers connect to their service provider. Vendor implementations of FQ typically classify packets into 256, 512, or 1024 queues using a hash function that is calculated across the source/destination address pair, the source/destination UDP/TCP port numbers, and the IP ToS (Typ of Service) byte. FQ requires minimal configuration (it is either enabled or disabled) and each of the n active queues is allocated $1/n$ of the output port bandwidth. As the number of queues changes, the bandwidth allocated to each of the queues changes. For example, if the number of queues increases from n to $(n + 1)$, then the amount of bandwidth allocated to each of the queues is decreased from $1/n$ of the output port bandwidth to $1/(n + 1)$ of the output port bandwidth.

FQ provides excellent isolation for individual traffic flows because, at the edges of the network, a typical subscriber has a limited number of flows, so each flow can be assigned to a dedicated queue, or else a very small number of flows, at most, are assigned to each queue. This reduces the impact that a single misbehaving flow can have on all of the other flows traversing the same output port.

In class-based FQ, the output port is divided into a number of different service classes. Each service class is allocated a user-configured percentage of the output port bandwidth. Then, within the bandwidth block allocated to each of the service classes, FQ is applied. As a result, all of the flows assigned to a given service class are provided equal shares of the aggregate bandwidth configured for that specific service class.

6.3.6 Earliest Due Date

In the Earliest Due Date (EDD) or Earliest Deadline First (EDF) scheduling scheme, a deadline is associated with each packet and the scheduler attempts to provide the target QoS metric by scheduling the packet transmission in the increasing order of the deadlines [Jeffay91, Saito90]. It is inspired in the earliest due date service discipline of the simulation theory, which assigns a deadline to each client and serves the clients by ascending order of deadlines. Typically, the packet deadlines are associated with their maximum tolerable delay. The

EDD scheduler assigns each packet with a deadline, which is the sum of its arrival time and the maximum tolerable delay, as expressed in Equation 6.7.

$$Deadline = a_i^k + d_k \tag{6.7}$$

where

a_i^k is the arrival time of the ith packet of session k,
d_k is the maximum delay bound for session k.

The earlier the deadline is, the earlier the packet can be served, and hence the higher priority the packet is assigned to get service, so packets are actually differentiated by priorities in terms of their deadlines. The EDD scheme arouses researchers interests due to its desirable property of being able to minimize the maximum lateness of packets [Guerin99], and as a result, throughput is maximized if late packets are discarded in real-time communications.

An important property of an EDD scheduler is that in a homogeneous traffic scenario the probabilities of deadline violations due to congestion are equal in all classes. This property holds independent of the total load and the load proportion of the different classes.

An additional option in operating an EDD scheduler is to discard packets that exceed their deadlines before entering service. This leads to a scheduling mechanism called Shortest Time to Extinction (STE)[Panwar88]. The STE policy is, e.g., appropriate for voice packets which become useless if they do not reach their destination within a certain time interval.

6.4 Advanced Service Disciplines

A large number of new scheduling disciplines have been proposed during the past several years in the literature, which are basically the variants of two fundamental disciplines, Generalized Processor Sharing (GPS) and Earliest Due Date (EDD) (or Earliest Deadline First (EDF)). In this section we will review the current scheduling schemes and outline their properties.

6.4.1 Weighted Round Robin

Weighted Round Robin (WRR) is the foundation for a class of queue scheduling disciplines that are designed to address the limitations of the FQ and PQ models.

- Weighted Round Robin addresses the limitations of the FQ model by supporting flows with significantly different bandwidth requirements. With WRR queuing, each queue can be assigned a different percentage of the output port's bandwidth.
- Weighted Round Robin addresses the limitations of the strict PQ model by ensuring that lower-priority queues are not denied access to buffer space and output port bandwidth. With WRR queuing, at least one packet is removed from each queue during each service round.

WRR is the simplest emulation of GPS, which serves a packet one time from each nonempty connection queue, instead of an infinitesimal amount (Figure 6.8).

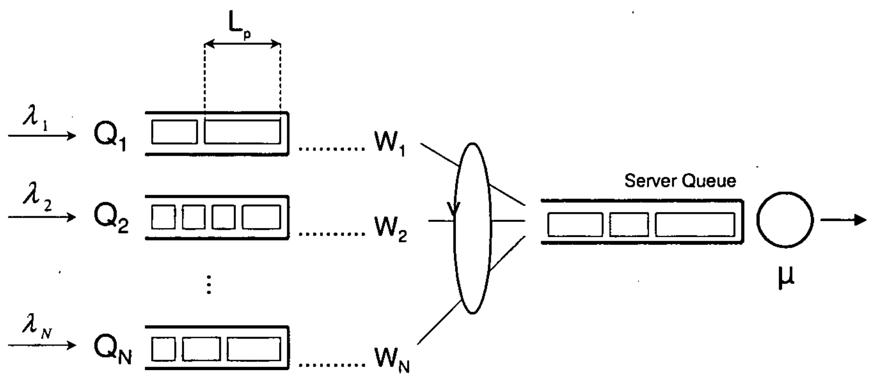

Figure 6.8: Weighted round robin

The service rate of session i is defined by:

$$r_i = \mu \cdot \frac{L_p}{\sum activeQueues} \cdot \frac{W_i}{\sum_{j \epsilon B_{GPS}(t)} W_j} \tag{6.8}$$

Round Robin (RR) approximates GPS reasonably well when all connections have equal weights and all packets have the same size. If connections have different weights, Weighted Round Robin serves a connection in proportion to its weight [Katevenis91]. If packets from different connections have different sizes, a WRR server divides each connection's weight by its mean packet size to obtain a normalized set of weights. However, in practice, a source's packet size may be unpredictable, so a WRR server cannot allocate bandwidth fairly.

In Weighted Round Robin queuing, packets are first classified into various service classes (for example, real-time, interactive, and file transfer) and then assigned to a queue that is specifically dedicated to that service class (like in the class based fair queuing approach). Each of the queues is serviced in a round-robin order. Similar to strict Priority Queuing and Fair Queuing, empty queues are skipped.

WRR queuing supports the allocation of different amounts of bandwidth to different service class by either:

- Allowing higher-bandwidth queues to send more than a single packet each time that it is visited during a service round, or
- Allowing each queue to send only a single packet each time that it is visited, but to visit higher-bandwidth queues multiple times in a single service round.

To regulate the amount of network resources allocated to each service class, a number of parameters can be tuned to control the desired behavior of each queue:

- The amount of *delay* experienced by packets in a given queue is determined by a combination of the rate that packets are placed into the queue, the depth of the queue, the amount of traffic removed from the queue at each service round, and the number of other service classes (queues) configured on the output port.
- The amount of *jitter* experienced by packets in a given queue is determined by the variability of the delay in the queue, the variability of delay in all of the other queues, and the variability of the interval between service rounds.

- The amount of *packet loss* experienced by each queue is determined by a combination of the rate that packets are placed into the queue, the depth of the queue, the aggressiveness of the RED profiles configured for the queue, and the amount of traffic removed from the queue at each service round. The fill rate can be controlled by performing traffic conditioning at some upstream point in the network.

WRR Queuing Benefits and Limitations

Weighted Round Robin queuing includes the following benefits:

- It can be implemented in hardware, so it can be applied to high-speed interfaces in both the core and at the edges of the network.
- It provides coarse control over the percentage of output port bandwidth allocated to each service class.
- It ensures that all service classes have access to at least some configured amount of network bandwidth to avoid bandwidth starvation.
- It provides an efficient mechanism to support the delivery of differentiated service classes to a reasonable number of highly aggregated traffic flows.

Additional the classification of traffic by service class provides more equitable management and more stability for network applications than the use of priorities or preferences. For example, if real-time traffic is assigned strict priority over file-transfer traffic, then an excessive amount of real-time traffic can eliminate all file-transfer traffic from the network. Weighted Round Robin queuing is based on the belief that resource reduction is a better mechanism to control congestion than resource denial. Resource denial not only blocks all traffic from lower-priority service classes but also obstructs all signaling regarding the denial of resources. As a result, TCP applications and externally clocked UDP applications are unable to correctly adapt their transmission rates to respond to the denial of network resources.

The primary limitation of Weighted Round Robin queuing is that it provides the correct percentage of bandwidth to each service class only if all of the packets in all of the queues are the same size or when the mean packet size is known in advance. Simple simulation have been done (see Figure 6.3) to show this limitation in a graphic. Besides, WRR scheduling is fair only over time scales longer than a round time. At a shorter time scale, some connections may get more service than others.

WRR Implementations and Applications

Because the WRR scheduling discipline can be implemented in hardware, it can be deployed in both the core and at the edges of the network to arbitrate the weighted distribution of output port bandwidth among a fixed number of service classes. Weighted Round Robin effectively overcomes the limitations of Fair Queuing by scheduling service classes that have different bandwidth requirements. Weighted Round Robin also overcomes the limitations of strict Priority Queuing by ensuring that lower-priority queues are not bandwidth-starved. However, WRR's inability to support the precise allocation of bandwidth when scheduling variable-length packets is a critical limitation that needs to be addressed.

6.4.2 Deficit Round Robin

Deficit Round Robin (DRR) queuing was proposed by M. Shreedhar and G. Varghese in 1996 [Shreedhar95]. Deficit Round Robin is the basis for a class of queue scheduling disciplines that are designed to address the limitations of the Weighted Round Robin and Weighted Fair Queuing models.

Deficit Round Robin modifies WRR to allow it to handle variable packet sizes without knowing the mean packet size of each connection in advance [Shreedhar95]. A Deficit Round Robin scheduler associates each connection with a deficit counter initialized to zero. The scheduler visits each connection in turn and tries to serve one quantum worth of bits from each visited connection. The packet at the head of the queue is served if it is not larger than the quantum size. If it is larger, the quantum is added to the connection's deficit counter. If the scheduler visits a connection such that the sum of the connection's deficit counter and the quantum is larger than or equal to the size of the packet at the head of the queue, the packet at the head of the queue is served, and the deficit counter is reduced by the packet size. The main attraction of Deficit Round Robin is its easy implementation. However, like Weighted Round Robin, it is unfair at time scales smaller than a round time.

- DRR addresses the limitations of the WRR model by accurately supporting the weighted fair distribution of bandwidth when servicing queues that contain variable-length packets.
- DRR addresses the limitations of the WFQ model by defining a scheduling discipline that has lower computational complexity and that can be implemented in hardware.

This allows DRR to support the arbitration of output port bandwidth on high-speed interfaces in both the core and at the edges of the network. In DRR queuing, each queue is configured with a number of parameters:

- A *weight* that defines the percentage of the output port bandwidth allocated to the queue.
- A *deficit counter* that specifies the total number of bytes that the queue is permitted to transmit each time that it is visited by the scheduler. The deficit counter allows a queue that was not permitted to transmit in the previous round (because the packet at the head of the queue was larger than the value of the deficit counter) to save transmission "credits" and use them during the next service round.
- A *quantum of service* that is proportional to the weight of the queue and is expressed in terms of bytes. The deficit counter for a queue is incremented by the quantum each time that the queue is visited by the scheduler.

If

$$quantum[i] = 2 * quantum[x],$$

then queue i will receive twice the bandwidth as queue x when both queues are active.

In the classic Deficit Round Robin algorithm, the scheduler visits each non-empty queue and determines the number of bytes in the packet at the head of the queue. The variable *DeficitCounter* is incremented by the value quantum. If the size of the packet at the head of the queue is greater than the variable *DeficitCounter*, then the scheduler moves on to service the next queue. If the size of the packet at the head of the queue is less than or equal to the variable *DeficitCounter*, then the variable *DeficitCounter* is reduced by the number of bytes in the packet and the packet is transmitted on the output port. The scheduler continues to dequeue packets and decrement the variable *DeficitCounter* by the size of the transmitted packet until either the size of the packet at the head of the queue is greater than the variable *DeficitCounter* or the queue is empty. If the queue is empty, the value of *DeficitCounter* is set to zero. When this occurs, the scheduler moves on to service the next non-empty queue.

DRR Benefits and Limitations

The benefits of Deficit Round Robin queuing are:

- Provides protection among different flows, so that a poorly behaved service class in one queue cannot impact the performance provided to other service classes assigned to other queues on the same output port.
- Overcomes the limitations of WRR by providing precise controls over the percentage of output port bandwidth allocated to each service class when forwarding variable-length packets.
- Overcomes the limitations of strict PQ by ensuring that all service classes have access to at least some configured amount of output port bandwidth to avoid bandwidth starvation.
- Implements a relatively simple and inexpensive algorithm that, from a computational perspective, does not require the maintenance of a significant amount of per-service class state.

As with other models, Deficit Round Robin queuing has limitations:

- Highly aggregated service classes mean that a misbehaving flow within a service class can impact the performance of other flows within the same service class. However, in the core of a large IP network, routers are required to schedule aggregate flows, because the large number of individual flows makes it impractical to support per-flow queue scheduling disciplines.

- DRR does not provide end-to-end delay guarantees as precise as other queue scheduling disciplines do.
- DRR may not be as accurate as other queue scheduling disciplines. However, over high-speed links, the accuracy of bandwidth allocation is not as critical as over low-speed links.

DRR Implementations and Applications

Because the Deficit Round Robin queue scheduling discipline can be implemented in hardware, it can be deployed in both the core and at the edges of the network to arbitrate the weighted distribution of output port bandwidth among a fixed number of service classes. Deficit Round Robin provides all of the benefits of WRR, while also addressing the limitations of WRR by supporting the accurate allocation of bandwidth when scheduling variable-length packets.

6.4.3 Hierarchical Round Robin

The Hierarchical Round Robin (HRR) service discipline was proposed by C. Kalmanek, H. Kanakia, and S. Keshav [Kalmanek90]. It is based on the definition of several hierachical service levels, each of which is serviced in round robin during a pre-determined number of slots. The higher the hierarchical level the greater the number of slots assigned to that level and, thus, the greater the bandwidth assigned to the flows belonging to the level. As all hierarchical levels are serviced during each cycle of the round-robin, the HRR service discipline guarantees limited delays, according to the position of flow in the service hierarchy.

6.4.4 Weighted Fair Queuing

Weighted-Fair-Queuing (WFQ) [Demers89] is the most well-known approximation of GPS scheduling, which do not make the infinitesimal packet size assumption of GPS and works with variable size packets, so it do not need to know a connections mean packet size in advance.

In WFQ, when the server is ready to transmit the next packet at time t, it picks, among all the packets queued in the system at t, the first packet that would complete service in the corresponding GPS system. By this we have to calculate the virtual time of the GPS system and the start and finish time of each packet in the virtual system using following equations:

$$VT_{(t)} = VT_{(t')} + \frac{\mu}{\sum_{k=1}^{N} W^k} \cdot (t - t') \tag{6.9}$$

$$S_i^k = max\{F_{i-1}^k; VT_{(t)}\} \tag{6.10}$$

$$F_i^k = S_i^k + \frac{P_i^k}{W^k} \tag{6.11}$$

In other words, the intuition behind WFQ is to compute the time a packet would complete service had we been serving packets with a GPS server, then serve packets in order of these finishing times (Figure 6.9).

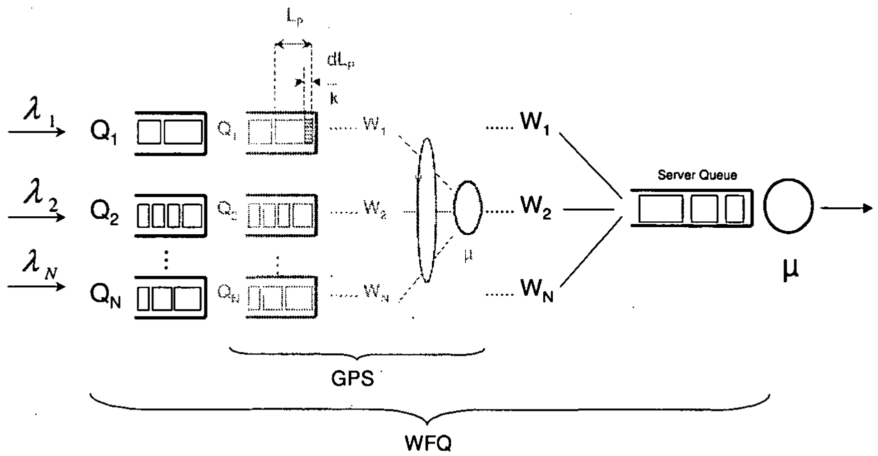

Figure 6.9: Weighted fair queuing

It is better to call a packet's finishing time under GPS a finish number, rather than a finish time, to emphasize that it is only a service tag that indicates the relative order in which the packet is to be served, and has nothing to do with the actual time at which the packet is served.

Even WFQ is known for a longer time and implemented in routers and various simulation environments, it is not easy to find a elaborate description to understand how WFQ really works. Because this is crucial to correct simulate WFQ we have done it by a step-by-step example in the performance evaluation part in the last chapter.

Weighted Fair Queuing Benefits and Limitations

The benefits of WFQ are:

- It supports different levels of QoS,
- Overcomes the starvation problem of priority queuing, and
- It is based on bandwidth rather than number of packets.

The drawback of WFQ is that it demands that a state is kept in the scheduler and therefore the implementation is complex.

6.4.5 Class Based Queuing

Class Based Queuing (CBQ) [Floyd95] is a traffic management algorithm developed by the Network Research Group at Lawrence Berkeley National Laboratory in the USA. It is an alternative to traditional router-based technology. Now in the public domain as an open technology, CBQ is deployed by companies at the boundary of their WANs.

Out of the previous section about FIFO we know that the FIFO queuing method can support many different types of applications - just not at the same time. For example, a low-latency service may be offered if the queue is always empty or nearly empty, or a high-bandwidth, moderate-loss service can be provided by a FIFO that is kept full by applications using a windowed flow control, such as TCP. Mechanisms such as Random Early Detection[2] (RED) attempt to achieve a middle ground by preventing the queue from becoming full by randomly dropping some of the arriving packets, but the result is a

[2]RED is a way to reduce the negative impact of a queue overflow. A weight can also be added to the packets in a RED drop mechanism, WRED. By doing so, the packets with a higher weight has a lower probability of being dropped.

compromise that satisfies few QoS requirements - especially as the number of flows sharing the FIFO queue increases.

Class-based queuing attempts to avoid this problem by sorting the traffic into different classes by examining the packet header (and sometimes the contents as well) and trying to determine the type of traffic to which the packet belongs. Once the packet has been classified, it is placed in a FIFO queue that contains only other packets of the same type. The queues of the different classes are then served according to established policies, whereby the priority queuing policy is the simplest one. Advanced methods are using feedback mechanisms composed of an estimator and a link sharing scheduler (Figure 6.10).

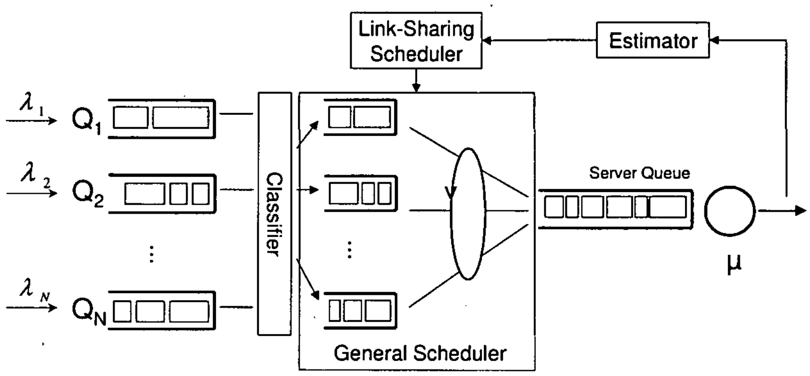

Figure 6.10: CBQ architecture

This allows the FIFO of each class to provide the desired type of service, but there are a also some problems with this approach. It requires a constant configuration burden because the operator has to configure the allocation of service to the different classes. If the network environment is dynamic, the class-based queuing method of allocating service could be impractical because the allocation is independent of the number of users of a given class. It provides less flexibility for delivering premium services.

CBQ lets classifying traffic in a multi-level hierarchy (Figure 6.11). For instance, two companies may first identify the overall needs of each department or business group, and then define the requirements of each application or group of applications within each department. For performance and architectural reasons, traditional router-based queuing schemes are limited to a small number of classes and only allow one-dimensional classification.

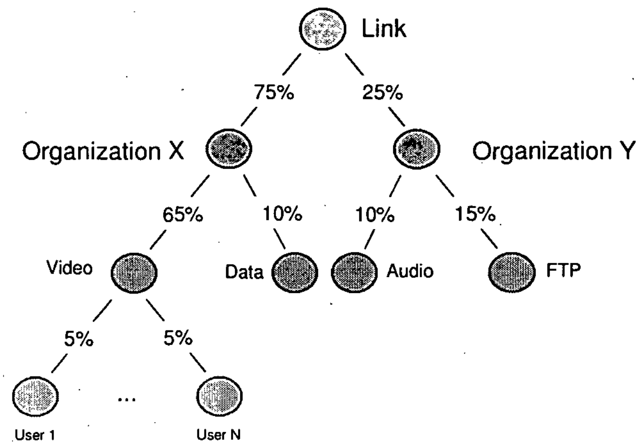

Figure 6.11: Multi-level hierarchy with CBQ

The class-based queuing approach assumes that it is possible to determine the type of traffic and therefore the type of service desired simply by looking at a packet. The theory is that the type of packet can be determined by simple properties of the packet, such as its port number. However, today many types of applications run over HTTP and there is no way to tell if a Web connection is carrying a business-critical, outsourced business application or a casual Web browsing session. As end-to-end encryption becomes more widespread (like in peer-to-peer VoIP clients as Skype[3]), it will become impossible to look inside the packet for information's to determine the class of the packet. Operators therefore lose traffic visibility when flows are encrypted, thus preventing the application of any meaningful QoS parameters.

Since encryption will only become more widespread, class-based queuing will become an even less optimal approach for QoS in the future. The limitations of class-based queuing and weighted RED (WRED) often result in the inappropriate discarding of packets. Since traffic flows are lined up in shared queues, it is impossible to isolate and discard those flows that are exceeding service level agreements guarantees before discarding traffic flows that are staying within their agreements. Packets are therefore discarded randomly from shared queues as they become full. The use of class-based queuing forces the network administrator to keep modifying packet classification configura-

[3] http://www.skype.com

tions to keep up with the latest popular Internet applications. However, the proliferation of new applications makes it highly impractical for operators to frequently adjust packet classifications to modify queuing methods.

6.4.6 Self-Clocked Fair Queuing

In order to reduce the complexity of WFQ for updating its virtual time on a packet arrival, an approximate implementation is proposed and analyzed in [Golestani94] under the name of Self-Clocked Fair Queuing (SCFQ). The SCFQ algorithm can achieve easier implementation as well as maintaining the fairness property by introducing a new virtual time function. The complexity in the WFQ scheduler arises from the fact that the scheduler defines fairness in the reference to the events in the hypothetical GPS scheduler, which creates the need for simulating events and computing the corresponding virtual time $VT(t)$.

The SCFQ scheme reduces the complexity by adopting a self-contained approach to the fairness definition. Similar to the WFQ scheduler, the SCFQ scheduler is also based on the notion of system's virtual time, viewed as the indicator of work progress in the system, except that the measure of virtual time here is found in the actual queuing system itself, rather than being derived from a hypothetical system. Unlike the extensive computations needed to evaluate $VT(t)$ in the WFQ algorithm (see equation 6.9), the virtual time $\widehat{VT}(t)$is simply extracted from the packet situated at the head of queue in the SCFQ algorithm instead, therefore the service tag is computed as:

$$F_k^i = \frac{L_k^i}{\phi_k} + max\{F_k^{i-1}, \widehat{VT}(a_k^i)\} \tag{6.12}$$

The major complexity of the SCFQ scheme is the maintenance of a sorted list of packets based on the corresponding service tags. Further work showed that there is also a large worst-case delay and short-term unfairness [Goyal97].

6.4.7 Start Time Fair Queuing

The computational benefits of SCFQ, but without the penalty of a large worst-case delay and short-term unfairness, can be obtained with a variant of SCFQ called Start Time Fair Queuing (STFQ) [Goyal97], in which both the start time (S_k^i) and the finish time (F_k^i) of each arriving packet are computed the same way as in WFQ (see Equations 6.10 and 6.11, respectively). The start time of an arriving packet at an inactive connection is set to the current virtual time. Otherwise, it is set to the finish time of its previous packet. A packet's finish time is the sum of its start time and its packet size divided by its weight. The virtual time $\widehat{VT}(t)$ is set to the start time of the packet currently in service. If there are no more packets to send, the virtual time is set to the largest finish time of any packets sent until that time. Packets are served in the increasing order of their start time. It can be shown that Start Time Fair Queuing has the same low implementation complexity as SCFQ, but with a much lower worst-case delay[Goyal97].

6.4.8 Worst-case Fair Weighted Fair Queuing

The Worst-case Fair Weighted Fair Queuing (WF^2Q) is proposed as another GPS approximation in [Bennett96], which calculates the service tag the same as WFQ in Equations 7.14 - 7.16. While WFQ uses only finish times of packets in the GPS system, WF^2Q uses both start times and finish times of packets in the GPS system to achieve a more accurate emulation. In WF^2Q, when the next packet is chosen for service at time t, rather than selecting it among all the packets at the server as in GPS, the server only considers the set of packets that have started (and possibly finished) receiving service in the corresponding GPS system at time t. Thereafter it selects the packet among them that will complete service first in the corresponding GPS system. The service provided by WF^2Q is almost identical to that of GPS, differing by no more than one maximum packet size. Compared to WFQ, WF^2Q solves the problem of having a possibly large gap between high weighted packets. Thus, while WF^2Q is fairer than WFQ, it is more complex than WFQ.

6.4.9 Stochastic Fairness Queuing

Because the WFQ scheduler needs to maintain the per-flow state information, attempts have been made to reduce the implementation complexity. One notable scheme that aims to approximate FQ at a smaller implementation cost is Stochastic Fairness Queuing (SFQ) proposed in [McKenny90]. SFQ classifies packets into a smaller number of queues than FQ by a hash function, so that the amount of state variables in SFQ are greatly reduced while its performance can approach to that of the original WFQ.

6.4.10 Virtual Clock

The Virtual Clock (VC) [Zhang90] discipline aims to emulate the Time Division Multiplexing (TDM). In short, each packet is allocated a virtual transmission time, which is the time at which the packet would have been transmitted were the server actually doing TDM. Packets are transmitted in the increasing order of virtual transmission times.

For better understanding, it should be noted that every flow has to have a specified average sending rate and to every flow a virtual clock is assigned (Figure 6.12). The virtual clock ticks at every packet arrival and stamps the packet by the flows virtual clock value. This value is based on the previously specified average sending rate. Afterwards the packet is queued in the common server queue in ascending order, as if the virtual clock stamp were the real time slot number in the TDM system.

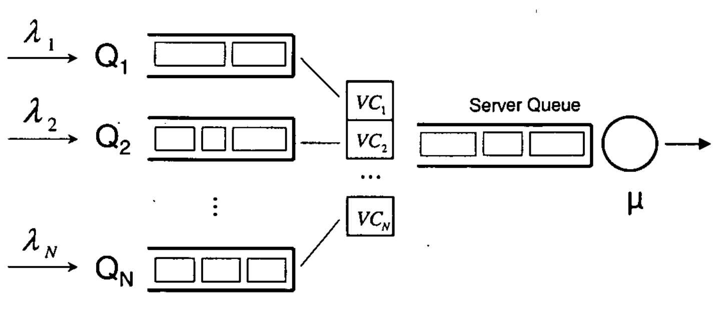

Figure 6.12: Virtual clock

The Virtual Clock algorithm orders packet transmission without changing the statistical sharing nature of packet switching – the network forwards all packets as long as resources are available. It also ensures that each well-behaving connection gets good performance.

The time tag is easier to compute as the virtual time in fair queuing algorithms as the Weighted Fair Queuing. However, this is at the expense of not having the same fairness property as the WFQ system. The algorithm for computing service tags has a few variants but the main idea is captured by the following equation:

$$F_i^k = \frac{L_i^k}{\phi_i} + max\{F_{i-1}^k, a_i^k\} \tag{6.13}$$

Comparing with Equations 6.10 and 6.11, the virtual time has been replaced with the real time a_i^k. When all connections are backlogged, Virtual Clock and WFQ provide identical service and identical worst-case end-to-end delay bounds [Goyal95]. Thus, Virtual Clock scheduling can substitute WFQ when serving guaranteed-service connections. However, when used for best-effort connections, the relative fairness bound for Virtual Clocks is infinity [Stiliadis96]. This means, when two connections are backlogged, one of them may obtain a throughput infinitely more than another. If only one scheduler had to be implemented in a switch than implementing WFQ is the better choice [Keshav97].

6.4.11 Core Stateless Fair Queuing

To reduce the cost of maintaining flow state information, a fair scheduling algorithm called Core Stateless Fair Queuing (CSFQ) is proposed in [Stoica98]. In this method, routers are divided into two categories – edge routers and core routers. An edge router keeps per flow state information and estimates each flow's arrival rate, and these estimates are inserted into the packet headers and passed on to the core routers. A core router simply maintains a stateless FIFO queue and drops a packet randomly based on the rate estimate in its header during the period of congestion. This scheme reduces the core router's design complexity but the edge router's design is still complicated. Besides, because of the estimated rate in the header, the core router has to extract packet information differently from traditional routers.

6.4.12 Delay Earliest Due Date

Delay Earliest Deadline First (Delay-EDD) is an extension of EDD where during the call setup the server negotiates a service contract with each source [Ferrari90]. The contract states that if a source obeys a peak and average rate descriptor, every packet on that connection receives a worse-case delay smaller than some bound. During the call admission, the scheduler ensures not only that the sum of the peak rates of the admitted calls is smaller than the link capacity, but also that even in the worst case, when every connection sends traffic at its peak rate, the scheduler meets its delay bound. Note that the delay bound for a connection is independent of its bandwidth reservation, in that a connection reserving a small bandwidth can still obtain a small delay bound by assigning a stringent deadline.

Therefore, unlike GPS-emulation schedulers, Delay-EDD separates the bandwidth and delay bounds, but at the cost of reserving bandwidth at the peak rate, which gives up the statistical multiplexing gain. Besides, the Delay-EDD scheduler also needs to buffer the per-connection state variable called Expected Deadline (ExD) similar to WFQ, which is defined with

$$ExD_i^k = max\{a_i^k + d_k, ExD_{i-1}^k + Xmin_k\} \tag{6.14}$$

where $Xmin_k$ is the minimum packet inter-arrival time for session k, and it is decided by the session peak rate. Thus its implementation is as complex as WFQ implementation, except that it does not need the virtual time computation.

6.4.13 Jitter Earliest Due Date

The Jitter-EDD discipline [Verma91] extends Delay-EDD to provide delay-jitter bounds (that is, a bound on the minimum as well on the maximum delay). After a packet has been served at each server, it is stamped with difference between its deadline and actual finishing time. A regulator at the entrance of the next switch holds the packet for this period before it is made eligible to be scheduled. This provides the required minimum and maximum delay guarantees.

6.4.14 Earliest Due Date for Finite Buffer

The Earliest Due Date for Finite Buffer (EDD-FB) is proposed to take the loss performance into consideration together with the delay performance [Lai98] for real-time traffic. The EDD-FB scheme cannot only maintain the desirable property for the EDD scheme to minimize the maximum lateness of packets, but also reduce the loss probability at the same time by considering the queue length information when the scheduler makes scheduling decisions. In the EDD-FB scheme, packets from each source are queued in a separate buffer for service. In addition, a threshold is set for each queue to determine whether the queue is in danger of overflow or not. When the occupancy of one or more queues is above the threshold, the scheduler will pick up the head-of-line packet from the longest queue and serve it. In this way, those queues that are in danger of overflow will get service first and therefore be able to recover quickly from the temporary overloading. However, the EDD-FB scheduler maintains an individual buffer for each connection, so the implementation complexity is very high, and the longest queue first strategy when several queues are in danger of overflow at the same time, results in the coherence in terms of the loss performance of the connections even if they may have different loss requirements.

6.4.15 Service Curve based Earliest Due Date

The Service Curve based Earliest Due Date (SCEDD) is proposed to provide guaranteed service for a session based on a flexible service specification function for each session called service curve in [Sariowan99]. The function is a generalized policy to which well-known policies such as Virtual Clock and the Earliest Deadline First can be mapped as special cases by an appropriate specification of the service curve. Based on the service curve, the SCEDD scheduler calculates the packet deadline and sorts packet transmission by its deadline. With the key capability for SCEDD to allocate and guarantee the service curve with arbitrary shapes, the SCEDD scheduler decouples the delay and bandwidth allocation. However, the deadline assignments under SCEDD are much more complex than the original EDD, and it is not clear whether it is feasible for all the network routers to maintain the same service curves for traffic classes.

6.5 Attributes on Analysis of Traffic Scheduling Algorithms

For the application of schedulers in real networks, it is important to analyze the characteristics of the service offered to individual sessions in a network of servers where the schedulers on the path of the session may use different scheduling [Stiliadis96]. Three important attributes are pointed out. These are (i) delay behavior, (ii) fairness, and (iii) implementation complexity.

6.5.1 End-to-End Delay Guarantees

In order to provide a deterministic delay bound, it is necessary to bound the burstiness of the session at the input of the network. The most common approach for bounding the burstiness of input traffic is by shaping through a token bucket [Turner86]. Several studies have used this traffic model [Cruz91a, Cruz91b, Parekh92b, Parekh94].

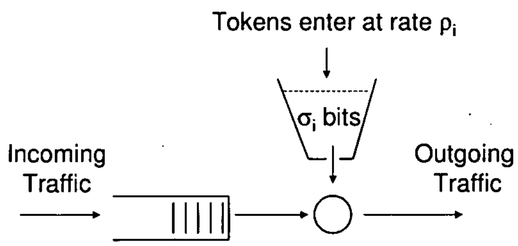

Figure 6.13: A token bucket

Consider the token bucket scheme of Figure 6.13. Tokens or permits are generated at a fixed rate, ρ, and packets can be released into the network only after removing the required number of tokens from the token bucket. There is no bound on the number of packets that can be buffered, but the token bucket contains at most σ bits worth of tokens. In addition to securing the required number of tokens, the traffic is further constrained to leave the bucket at a maximum rate of $C \geq \rho$.

The constraint imposed by the token bucket is as follows: If $A_i(t, t')$ is the amount of session i flow that leaves the token bucket and enters the network in the time interval $(t, t']$, then

$$A_i(t,t') \leq \min\{(t'-t)C_i, \sigma_i + \rho_i(t'-t)\}, \forall t' \geq t \geq 0, \tag{6.15}$$

for every session i.

We say that session i conforms to (σ_i, ρ_i, C_i), or $A_i \sim (\sigma_i, \rho_i, C_i)$ [Parekh92b]. The arrival constraint is attractive since it restricts the traffic in terms of average rate (ρ), peak rate (C), and burstiness (σ and C).

In deriving the end-to-end delay bound for a particular session, however, we do not make any assumptions about the traffic from the rest of the sessions sharing the same links of the network. In addition to minimizing the end-to-end delay in a network of servers, the delay behavior of an ideal algorithm includes the following attributes:

- Insensitivity to traffic patterns of other sessions: Ideally, the end-to-end delay guarantees for a session should not depend on the behavior of other sessions. This is a measure of the level of isolation provided by the scheduler to individual sessions. Note that isolation is necessary even when policing mechanisms are used to shape all the flows at the entry point of the network, as the flows may accumulate burstiness within the network.
- Delay bounds that are independent of the number of sessions sharing the outgoing link: This is necessary if the algorithm is to be used in switches supporting a large number of flows.
- Ability to control the delay bound of a session by controlling only its bandwidth reservation: This property of the algorithm provides significant flexibility in trading off session delays with their bandwidth allocations.

6.5.2 Fairness

Significant discrepancies may exist in the service provided to different sessions over the short term among scheduling algorithms. Some schedulers may penalize sessions for service received in excess of their reservations at an earlier time. Thus, a backlogged session may be starved until others receive an equivalent amount of normalized service, leading to short-term unfairness. Therefore, two scheduling algorithms capable of providing the same delay guarantee to a session may exhibit vastly different fairness behaviors.

While there is no common accepted method for estimating the fairness of a scheduling algorithm, it is easy to define fairness in an informal manner. In general, we would like the system to always serve connections proportional to their reservations and distribute the unused bandwidth left behind by idle sessions equally among the active ones. In addition, sessions should not be penalized for excess bandwidth they received while other sessions were idle. Following Golestani's work [Golestani94], we define the fairness parameter of a scheduling algorithm as the maximum difference between the normalized service received by two backlogged connections over an interval in which both are continuously backlogged.

Based only on the end-to-end delay bounds and fairness properties, Generalized Processor Sharing (GPS) is an ideal scheduling discipline. As previously shown, GPS multiplexing is defined with respect to a fluid-model, where packets are considered to be infinitely divisible. Thus, GPS serves each backlogged session with a minimum rate equal to its reserved rate at each instant; in addition, the excess bandwidth available from sessions not using their reservations is distributed among all the backlogged connections at each instant in proportion to their individual reservations. This results in perfect isolation, ideal fairness, and low end-to-end session delays. The only problem is that GPS is not implementable in real systems because packets are not infinitely divisible and the server cannot serve backlogged session with a minimum rate. The packets can only be served one-by-one.

The Virtual Clock algorithm, in contrast, does not bound the difference in service received by two backlogged sessions over an interval that is smaller than the backlogged period. This is the result of the scheduler performing an averaging process on the rate of service provided to individual sessions. In Virtual Clock, the averaging interval can be arbitrarily long. The GPS scheduler, on the other hand, occupies the opposite extreme where no memory of past bandwidth usage of sessions is maintained. Note that, according to the presented definition of fairness, some amount of short-term unfairness between sessions is inevitable in any packet-level scheduler, since each packet must be serviced exclusively. In practice, we can only require that the difference in normalized service received by two sessions be bounded by a constant.

6.5.3 Implementation Complexity

Finally, schedulers differ greatly in their implementation complexity. The scheduling algorithm may need to be implemented in hardware in a high-speed network. In addition, it is desirable to have the time-complexity of the algorithm not depend on the number of active connections in the scheduler.

If V is the maximum number of connections that may share an output link, the implementation of a scheduler based on the sorted-priority architecture involves three main steps for processing each packet [Zhang91]:

- Calculation of the timestamp: The WFQ scheduler has the highest complexity in this respect, since a GPS scheduler must be simulated in parallel in order to update the virtual time. This simulation may result in a process overhead of $O(V)$ per packet transmission in the worst-case. On the other hand, in both Virtual Clock and Self-Clocked Fair Queuing the timestamp calculation involves only a constant number of computations, resulting in a worst-case complexity of $O(1)$.
- Insertion in a sorted priority list: The first packet of each session's queue must be stored in a sorted priority list. When a packet arrives into an empty queue, its insertion into the priority list requires $O(logV)$ steps.
- Selection of the packets with the minimum timestamp for transmission: Since the packets are stored in a sorted-priority structure, the packet with the highest priority may be retrieved in $O(logV)$ time [Cormen90].

Frame-based algorithms such as Weighted Round Robin (WRR) and Deficit Round Robin (DRR) can be implemented in $O(1)$ time, without any timestamp calculations. Unfortunately, these algorithms yield delay bounds that may grow linearly with the number of sessions sharing the outgoing link. Thus, in practice, the scheduling algorithm must trade-off the complexity of implementation with the other desirable properties of low delay and bounded short-term unfairness.

6.5.4 Performance vs. Implementation Complexity

As we have seen in the previous sections about end-to-end delay, fairness and implementation complexity, there is a trade off between performance and the implementation complexity of a scheduling algorithm. Recently there has been shown that there is a fundamental tradeoff between the delay bound and its computational complexity [Xu02]. They introduce the concept of GPS-relative delay (the difference between the time a packet finish service in a packet scheduling algorithm and its virtual finish time under GPS). For example, WFQ-like algorithms are $O(1)$ GPS-relative while round-robin ones are $O(n)$ GPS-relative. It is shown that $\Omega(log(n))$ is the lower bound of computational complexity to provide GPS-relative delay. Furthermore, any point between $O(1)$ and $O(n)$ GPS-relative delay also requires a $\Omega(log_2 n)$ lower bound complexity. That is to say, there is no such algorithm that can be as simple as Round Robin but possess the delay properties as good as WFQ.

The question of choosing the proper scheduling algorithm in switch or router design is not a simple one. One answer could be to choose the algorithm with the best performance but as we have seen this is associated with higher complexity. Another possibility is to choose the one with the least cost that satisfies predefined performance goals. Traditionally, schedulers are compared under their worst-case conditions [Stiliadis95]. The rationale behind this approach is that guaranteed services are concerned mainly with upper bounds on delay rather than the average delay that a schedular can provide to a specific flow. As we have seen in Section 6.5.1, if a router implements a WFQ scheduling discipline, and the traffic can be characterized by a token bucket then there will be an absolute upper bound on the network delay of the traffic in question.

Research has been done also from a more balanced perspective. In [Wangdong03] the DRR and WF^2Q algorithms are compared, even though WF^2Q guarantees delay bounds and DRR does not. This is true if this two algorithms are compared under the same condition but it is not necessarily true if the DRR (or any other frame based schedular) has more bandwidth to use than the WF^2Q scheduler. A key observation of this work is that with enough additional bandwidth round robin schedulers can match any sorted-priority schemes in delay performance.

7 Performance Analysis

7.1 Introduction

We have already worked out the problem of congestion in the introduction part. We defined congestion as the state of sustained network overload where the demand for network resources is close to or exceeds capacity. By this, performance and transmission quality of the network can significantly be deteriorated.

The quality of a telephone conversation is assured in case of circuit switching (Time Division Multiplexing, TDM). Circuit switching naturally implies the use of admission control - new calls may get a busy tone if they arrive when a network link is congested. Admission control can be applied selectively to different classes of calls, so that the lowest-priority demands are rejected when just a small number of channels are still free. This protects the higher-priority calls, which are blocked only when the very last channel is busy.

On the other hand, IP networks suffer from the problem of congestion because this is inherent in best effort packet networks due to uncoordinated resource sharing. It is possible for several IP packets to arrive at the router simultaneously, needing to be forwarded on the same output link. To solve this problem one solution would be the introduction of QoS mechanisms like scheduling and complete architectures, either detailed worked out in previous chapters. Another solution is to over provision the network capacity.

The telephone network is over provisioned to a degree determined by the target busy-hour blocking probability, typically around one blocked call per thousand. This is low enough to make the network virtually transparent, as opposed to other causes of call failure, such as called-party busy or absent. Internet Service Providers (ISP) today are deploying overprovisioning because it seems to be the lowest-cost solution because of the huge amounts of bandwidth in backbones as a result of advances in Wavelength Division Multiplexing (WDM) technology, which has increased per-fiber bandwidth hugely. Studies have shown that in most big ISPs, if they do have a congestion problem in the network itself, generally it is a very small part that is affected. Adding bandwidth to that small part is a lot a less complex than putting in specific network wide mechanisms to provide QoS and MPLS-type traffic engineering.

Most of the Tier-1 ISPs indicate that they over provision their networks and generally engineer it such that they will not lose packets inside their own backbone. The recurring operations costs (e.g., network operation staff, deployment staff and troubleshooting staff) are believed to be much higher than the capital costs of building a network, especially if the ISP already has dark fibres facilities in place. This belief might relate to the current lack of widely deployed QoS in large backbone networks. As to the costs of overprovisioning, once one has access to dark fibre, it is not particularly more expensive (in capital costs) to light it with Gigabit Ethernet than 10 or 100 Megabit Ethernet.

A common capacity planning objective is 100% overprovisioning compared to peak demand. That is, the network has twice as much capacity as the expected demand in the busiest period of the day. This is done mainly for reasons of availability, as the network should be able to handle demand even under a link or node failure. In practice, overprovisioning levels tend to be much higher than this, with link utilization levels as low as 20% or even less.

It should be hold in mind that what we are talking about is the development of new services in third generation mobile networks. The trend is replacing TDM by IP technology but IP technology should not be equated with the Internet. The Internet is public; the IP backbone of a mobile operator is not. What we get by switching from TDM to IP is the congestion problem and the Internet (or better to say the Internet community) is giving valuable information on how to handle this problem. In this work the focus is on QoS mechanisms like scheduling and different QoS architectures, but because of currently predominant use of overprovisioning as a solution to the congestion problem in the Internet, simulations are carried out to investigate the feasibility of this approach.

7.2 Simulation Model

In order to study feasibility of overprovisioning the links, some simulations have been conducted. In this simulation model, the link overprovisioning concept and the QoS controlled concept are compared, and have the following relationships:

Concept	Feature	QoS Level
Link overprovisioning	$C + C'$	x
QoS Control	$C + QoSMechanisms$	x

Table 7.1: Link overprovisioning and QoS concept

In order to provide a QoS level x, link overprovisioning requires additional capacity C'. However, QoS control can meet the same QoS level by use of a simple QoS control mechanism with two classes – high and low priority packets. The model is depicted in Figure 7.1.

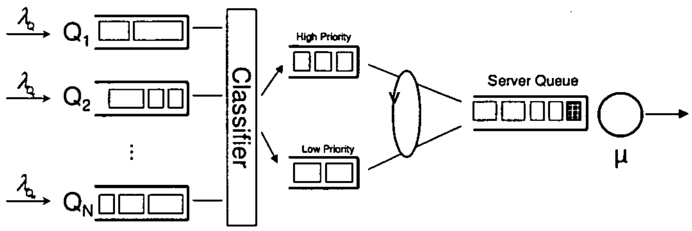

Figure 7.1: QoS control simulation model

The classifier in Figure 7.1 classifies all the incoming packets into these two types. For the QoS controlled concept two different scheduling mechanisms are included – Class Based Queuing (CBQ) and Weighted Fair Queuing (WFQ). In the classical approach of Class Based Queuing the server operates in the strict priority queuing mode by serving low priority packets only when there is no packet in the high priority queue. As long as there is a packet in the high priority queue, none of the low priority packets will be served. On the other hand, WFQ is a fair queuing discipline which provides protection to each service class by ensuring a minimum level of output bandwidth independent of the behavior of other service classes.

The main characteristic of the QoS model is that the percentage of high priority incoming traffic is varying between $10 - 90\%$ and the capacity of the outgoing link is fixed to 100 Mbit/s.

The link overprovisioning model does not make any difference between the packets and therefore it does not make any sense to vary the amount of high priority packets. The varying parameter in this case is the outgoing capacity. The main goal of this performance analysis, is to find out the trade-off between adding capacity to the outgoing link and prioritizing packets with QoS Mechanisms. The link overprovisioning model is depicted in Figure 7.2.

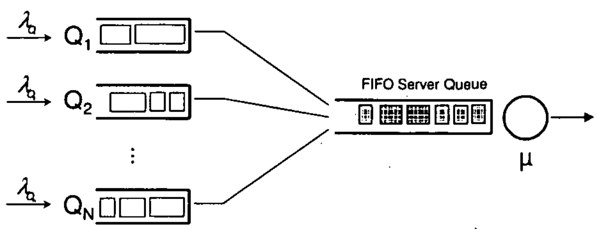

Figure 7.2: Overprovisioning simulation model

Parameters used in this simulations are summarized in Table 7.2.

Parameter	Value or Explanation
Input traffic amount	Over 90% of the link capacity
Priorities of packets	High and Low
Lengths of packets	Uniformly distributed between 21 and 1500 bytes
Input traffic distribution	Poisson distribution
Buffering	Store and forward
Transmission line speed	100 Mbit/s
Buffer size	Infinite

Table 7.2: Parameters of the simulation model

7.3 Modelling WFQ

Modelling the Weighted Fair Queuing algorithm is started with the example presented in this section. Out of this example, a flow chart is build, which is the basis for WFQ implementation in this simulation environment.

As mentioned in Section 6.4.4, concerning the calculation of start and finish time, it is better to use the notation of *start number* and *finish number*, respectively. Using the term "number" is better than using the term time, to emphasize that it is only a service tag that indicates the relative order in which the packet is to be served, and has nothing to do with the actual time at which the packet is served.

In this example, we suppose a system of three queues - A, B, and C. Let the server speed be 1 bit/s and at $t = 0$ the packets $P_1^A = 1$ bit, $P_1^B = 2$ bits, and $P_1^C = 2$ bits arriving as the first packets in queues A, B, and C, respectively. Later, at time $t = 4$ there will be another packet arriving at queue A, $P_2^A = 2$ bits (Figure 7.3). For simplicity, we suppose equal weights for all three queues. Later we will add the weight factors.

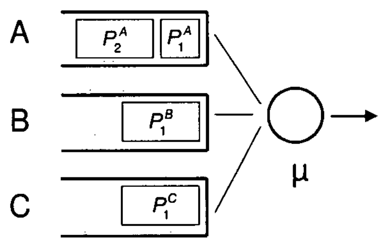

Figure 7.3: WFQ example

At time $t = 0$ the server has to take a packet out of one of the active queues, but the question is which one to choose - P_1^A, P_1^B, or P_1^C?

The WFQ algorithm will choose the packet with the smallest finish number but before being able to assign a finish number to a packet, the virtual time and the start number have to be calculated. Every calculation of these three parameters is triggered by the packet arrival in the WFQ system. A packet departure out of the WFQ system causes only the assignation of a new virtual time. It will be seen later that this recalculation of virtual time is the most complex part.

At time $t = 0$, the virtual time is $VT_{(t=0)} = 0$. The equation for calculating the virtual time is:

$$VT_{(t)} = VT_{(t')} + \frac{\mu}{N} \cdot (t - t') \tag{7.1}$$

where

$VT_{(t)}$ is the actual virtual time at time t,

$VT_{(t')}$ is the last update of the virtual time,

μ is the server rate (or line speed), and

N is the number of backlogged (or active) queues.

First of all, it is worth to point out again that the virtual time is not the real time of the system (the WFQ system). It is just a permanent increasing dimensionless number, sometimes also called round number, which indicates how many times the WFQ system goes through all active queues. As we can see out of Equation 7.1, the "speed" of the virtual time is reciprocally proportional with the number of active queues. This means if the number of active queues, N, is lower, the virtual time will go faster. If N is higher, the virtual time will go slower. The virtual time is required for calculating the start number. The start number is calculated out of the following equation:

$$S_i^k = max\{F_{i-1}^k; VT_{(t)}\} \tag{7.2}$$

where

S_i^k is the start number of the i^{th} packet of connection k,

F_{i-1}^k is the finish number of the $(i-1)^{th}$ packet of connection k, and

$VT_{(t)}$ is the virtual time at time t.

In Equation 7.2, the start number is either the finish number of the previous packet in this queue or the actual virtual time. In other words, if the packet arrives to an non-empty queue, it will be served after the last packet of this queue has been served (FIFO order inside the queues!). If the packet is the first packet in the queue (i.e., the packet is opening a new queue) then the start number of this packet will be equal to the actual virtual time. With the start number we can now calculate the finish number of the packet. The equation is as follows:

$$F_i^k = S_i^k + \frac{P_i^k}{N} \tag{7.3}$$

where

F_i^k is the finish number of the i^{th} packet of connection k,

S_i^k is the start number of the i^{th} packet of connection k,

P_i^k is the packet length of the i^{th} packet of connection k, and

N is the number of backlogged (active) queues.

The finish number is proportional to the packet length and reciprocally proportional to the number of backlogged queues. Later we will see that by substituting N whit the weight of the k^{th} queue, W^k, we get the equation for the weighted case of WFQ. Just to prerecord that if the weights are higher, the finish number is lower.

According to Equations 7.1 to 7.3, we can now calculate the finish numbers of the packets P_1^A, P_1^B, and P_1^C and they are $F_1^A = 1, F_1^B = 2$, and $F_1^C = 2$, respectively. P_1^A is the packet with the smallest finish number and it will be served first. Packets P_1^B and P_1^C have the same finish number of two. If the decision which one to take first is randomly it will not effect the long time fairness but a common solution is to first take the packet from the queue with the smaller index. In this sense P_1^B will be the second and P_1^C the third packet served.

Figure 7.4 visualize the server queue with the packet order according to the finish numbers, the virtual time of the GPS server, the WFQ system with there correspondent real-time axis and the packet in and out order in the WFQ system. This visualization will be of help in understanding the relationship between the virtual time and the real time and it will be shown on every change in the system, starting with this figure at time $t = 0$.

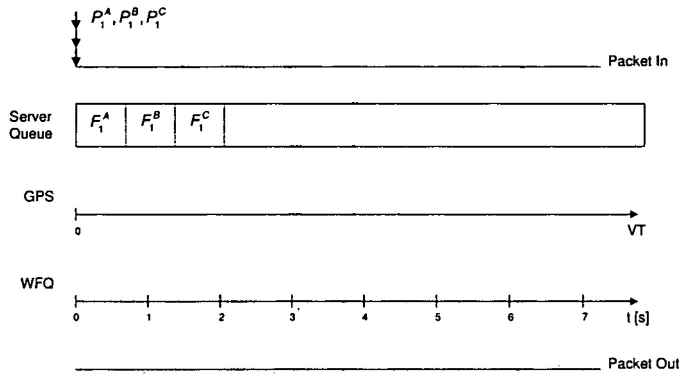

Figure 7.4: WFQ example at $t = 0$

Now it is important to point out that the order of the packets in the server queue, in WFQ depends only on the finish number of the packets by taking the packet with the smaller finish number at first. Every packet get assigned a finish number by entering the system and this finish number is not changing during time, independent of future arrivals to the system or the specific queue. This is one of the advantages of WFQ. The only thing that cannot be appointed in advance, is at which time the real-time WFQ system will serve the packet. By this, it is neither possible to know the finishing time of the packet unless the packet is leaving the system. Therefore, the notation of finish time and start time for WFQ is sometime confusing even when the prefix "virtual" has been used, and exactly this was the reason why we are using the notation of finish number and start number.

From above statement, it is not remarkable any more if we state that we do not know at what time the virtual time VT will be one and at what time it will be two.

At time $t = 1$, the first packet is leaving the system. On every packet departure and arrival, we have to do two things. First, we have to calculate the virtual time according to Equation 7.1. What we get is

$$VT_{(t=1)} = VT_{(t=0)} + \frac{1}{3} \cdot (1 - 0) = 0 + \frac{1}{3} \cdot 1 = \frac{1}{3} \tag{7.4}$$

The second thing that has to be done is to check if the smallest finish number at each backlogged queue is strictly greater of this virtual time to obtain a GPS queue activity check.

$$F^k_{min} > VT_{(t=1)} \tag{7.5}$$

In this case F^A_1, F^B_1, and F^C_1 are strictly greater than $VT_{(t=1)}$. Due to this, the previously calculated virtual time in Equation 7.4 is correct and so the results can be presented by extending the previous figure:

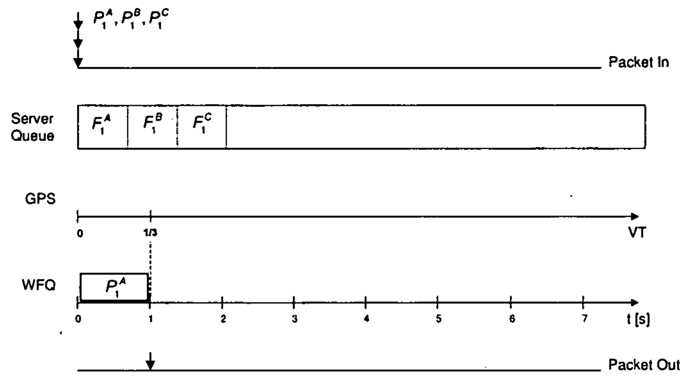

Figure 7.5: WFQ example at $t = 1$

The second step is necessary because it could happen that by finishing serving a packet in WFQ, this packet is already out of the GPS system. Thus, it is clear that packets in WFQ will not be finished later as the same packet in GPS and thereby the end-to-end delay bound of GPS is also guaranteed for WFQ. If at least one of the finishing numbers is smaller than the virtual time, we know that one packet already left the GPS system. If this packet is also the last packet of the appropriate queue and thereby the queue become inactive, the already calculated virtual time $VT_{(t)}$ is not correct anymore because of a wrong N. A new virtual time could cause other queues to become inactive. This problem is called the *iterated deletion problem* [Keshav97]. In this example, this is not the case but it will be added later on and it will be also shown how this reflects on the calculation.

By calculating the virtual time in Equation 7.4 we got $VT_{(t=1)} = \frac{1}{3}$ and a logical explanation is that in a GPS system with three backlogged queues the system will serve each of these queues with one-third of the server rate. In this example, the server rate is 1 bit/s and in this sense it needs three seconds to make a whole round. Analogously after one second it will be only at one-third of the full round.

The next event in the WFQ system will be the departure of packet P_1^B at time $t = 3$. The values we get, according to Equations 7.1 and 7.5 are:

$$\begin{aligned} VT_{(t=3)} &= VT_{(t=1)} + \frac{1}{3} \cdot (3-1) = \frac{1}{3} + \frac{1}{3} \cdot 2 = \\ &= 1 \end{aligned} \tag{7.6}$$

and

$$\begin{aligned} &F_1^A = 1 \not> 1 \\ &F_1^B = 2 > 1 \\ &F_1^C = 2 > 1 \end{aligned} \tag{7.7}$$

What we got now is a finish number not greater than the calculated virtual time. This means that packet P_1^A has already finished service in the GPS system. Because this was the only packet in queue A and the queue is become inactive, the number of backlogged queues is not correct anymore. In this way, also the calculated virtual time has to be recalculated. As explained before, the decrease of the number of backlogged queues will speed up the virtual time and by this generally it could happen that also other queues become inactive, causing an iterated deletion from the list of active queues. This is the main difficulty in WFQ, because computing the list of deletions has to be done upon any packet arrival and departure and is therefore computing intensive.

The exact time when packet P_1^A is finishing their service in GPS can be recursively calculated using Equation 7.1. The additional condition is

$$VT_{(t=t^*)} = F_1^A = 1 \tag{7.8}$$

By doing so we will find out that $t^* = 3$. In this case, the virtual time will not change, because $t^* = t$ but from this time we have to calculate with $N = 2$. The value of the virtual time can be calculated by Equation 7.1 for any value of t, but we are interested primary in values of VT for the packet arrival and

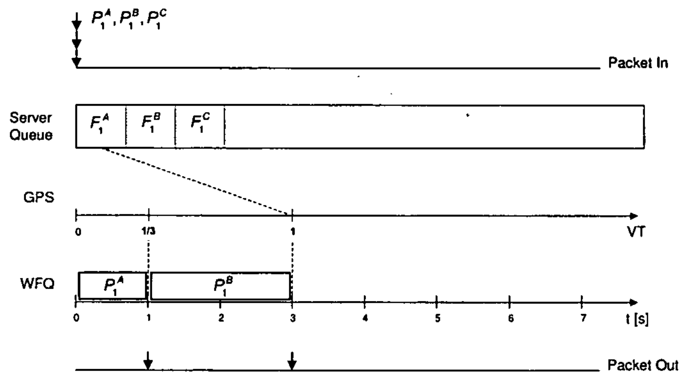

Figure 7.6: WFQ example at $t = 3$

departure in the WFQ system. The calculated values can be seen in Figure 7.6.

The next event will occur at time $t = 4$ by the arrival of packet $P_2^A = 2$. First, the virtual time has to be calculated:

$$\begin{aligned} VT_{(t=4)} &= VT_{(t=3)} + \frac{1}{2} \cdot (4-3) = 1 + \frac{1}{2} \cdot 1 = \\ &= 1\frac{1}{2} \end{aligned} \tag{7.9}$$

Now we can calculate the start and finish number for P_2^A. They are $S_2^A = 1\frac{1}{2}$ and $F_2^A = 3\frac{1}{2}$. From this time on, the number of active queues is again $N = 3$. By proofing, that F_{min}^k for all active queues is greater then the calculated virtual time the second step has been finished too. Now the graphical representation looks like follows:

According to the finish number, packet P_2^A is ordered behind the other packets in the priority queue and we already draw in the serving times for this packet in the real WFQ system.

At time $t = 5$, packet P_1^C is leaving the WFQ system. Calculating the virtual time we get

$$VT_{(t=5)} = VT_{(t=4)} + \frac{1}{3} \cdot (5-4) = 1\frac{1}{2} + \frac{1}{3} \cdot 1 =$$

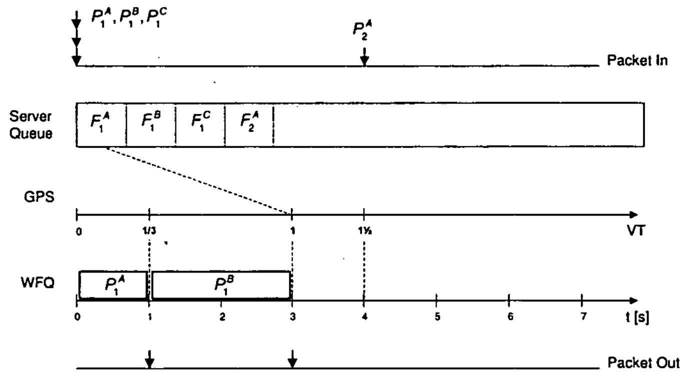

Figure 7.7: WFQ example at $t = 4$

$$= 1\frac{5}{6} \tag{7.10}$$

The queue activity check do not make any change. Figure 7.8 visualizes the progress.

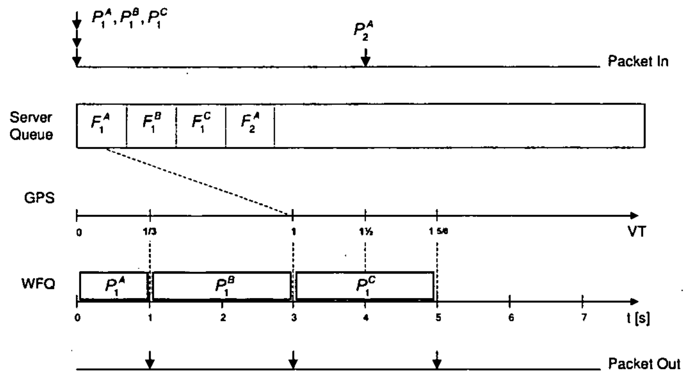

Figure 7.8: WFQ example at $t = 5$

The next event happens at time $t = 7$ with the departure of packet P_2^A. Now the virtual time is

$$VT_{(t=7)} = VT_{(t=5)} + \frac{1}{3} \cdot (7-5) = 1\frac{5}{6} + \frac{1}{3} \cdot 2 =$$

$$= 2\frac{2}{3} \tag{7.11}$$

What we get now by doing the queue activity check is that even two finish numbers are smaller then the actual virtual time (F_1^B and F_1^C). Because this two packets are the last packets in their corresponding queues, the virtual time is not exact. By doing the same recursive calculation as before, we get $t^* = 5.5$ and $N = 1$. Now we can write

$$VT_{(t=5.5)} = 2 \tag{7.12}$$

and present the result by Figure 7.9. According to Equation 7.12 we can

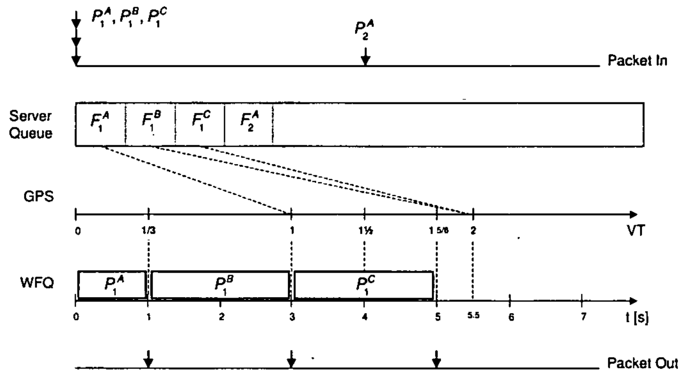

Figure 7.9: WFQ example at $t = 5.5$

calculate the exact virtual time for $t = 7$

$$\begin{aligned} VT_{(t=7)} &= VT_{(t=5.5)} + \frac{1}{1} \cdot (7 - 5.5) = 2 + \frac{1}{3} \cdot 1\frac{1}{2} = \\ &= 3\frac{1}{2} \end{aligned} \tag{7.13}$$

With this new virtual time, the queue activity check results in having also packet P_2^A finished service in the GPS system. The exact virtual time is again $VT_{(t=7)} = 3\frac{1}{2}$, because $t^* = t$. But if there is no packet in the GPS system, we can re-initialize the system by starting with $VT_{(t=0)} = 0$.

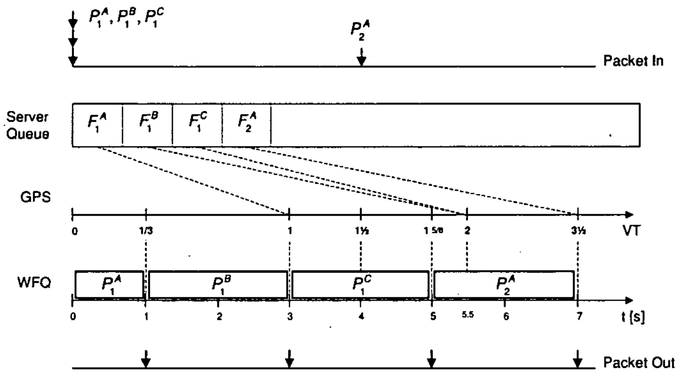

Figure 7.10: WFQ example at $t = 7$

To summarize - the order of the packets in the priority queue is defined by the increasing value of the finish number. The finish number is calculated at packet arrival time and it does not change, independent on future arrivals. What we need to calculate the finish number is the packet length and the virtual time. Because of the iterated deletion problem, we have to check the queue activity in the GPS system. In this approach, it is done on every packet arrival and departure in the WFQ system. In Figure 7.10, we can see that the virtual time runs faster if the number of active queues is smaller and it runs slower if the number of active queues is bigger.

7.3.1 Equations with Weighting

The example was given without taking in consideration different weights of the individual queues. If we assess the queues with appropriate weights, W^k, which are not changing over time, than the corresponding equation for calculating the virtual time, the start number and the finish number are:

$$VT_{(t)} = VT_{(t')} + \frac{\mu}{\sum_{k=1}^{N} W^k} \cdot (t - t') \tag{7.14}$$

$$S_i^k = max\{F_{i-1}^k; VT_{(t)}\} \tag{7.15}$$

$$F_i^k = S_i^k + \frac{P_i^k}{W^k} \tag{7.16}$$

respectively, where W^k is the weight of the k^{th} queue and $\sum_{k=1}^{N} W^k$ is the sum of the weights of all active queues.

7.3.2 WFQ Algorithm

By the above example, it is possible to define the WFQ algorithm. The main part is given in Figure 7.11. In the phase of initialization we define

$$\begin{aligned} VT_{(t=0)} &= 0 \\ N &= 1 \end{aligned} \tag{7.17}$$

where VT is the virtual time, and N the number of active queues at time $t = 0$.

The most important part of WFQ is always to have a correct virtual time. If the virtual time is not correct, incoming packets get a wrong finish number and by this WFQ cannot be fair any more. The virtual time depends on the number of active queues (N), whereby this number can change if the packet leaving the system is the last packet in its corresponding queue.

The simulation is a discrete event simulation. Each packet arrival and departure constitutes of one event. For each packet coming into the WFQ system the finish number is calculated according to

$$F_i^k = S_i^k + \frac{P_i^k}{W^k} \tag{7.18}$$

where

S_i^k is the start number of the i^{th} packet of connection k,

P_i^k is the packet length of the i^{th} packet of connection k, and

W^k is the weight of the k^{th} queue.

If the packet is the first packet in the queue, the start number S_i^k will be the actual virtual time calculated by

$$VT_{(t)} = VT_{(t')} + \frac{\mu}{\sum_{k=1}^{N} W^k} \cdot (t - t') \tag{7.19}$$

where

$VT_{(t')}$ is the last update of the virtual time,

μ is the server rate (or line speed), and

N is the number of backlogged (or active) queues.

If the packet is not the first one in the queue, the start number will be the finish number of the previous packet in this queue.

The finish number does not change over time and will be written in the first table (WFQ_In). For every packet leaving the WFQ system, the finish number is erased from the first table and written in the second one (WFQ_Out). For re-calculating the virtual time, only the second table is of relevance. The re-calculation process is triggered by a packet departure if any F^k_{min} out of the second table is smaller or equal to the virtual time:

$$F^k_{min} \leq VT_t \tag{7.20}$$

In this case, the affected F^k_{min} will be deleted from the second table and if it was also the last packet in its queue, the number of active queues N becomes now

$$N = N - 1$$

and by this the virtual time has to be re-calculated. First, the last update of the virtual time has to be re-calculated according to

$$VT_{t'} = VT_{t'} + \frac{\mu}{N} \cdot (t^* - t') \tag{7.21}$$

By this, the correct virtual time at time t will be

$$VT_t = VT_{t'} + \frac{\mu}{N} \cdot (t - t') \tag{7.22}$$

whereby t^* is the time at which the packet has left the system. This is calculated from $VT_{(t=t^*)} = F^k_{min}$, whereby it follows that

$$t^* = \frac{N}{\mu} \cdot (F^k_{min} - VT_{t'}) + t' \tag{7.23}$$

The new virtual time from Equation 7.22 will be greater than the old one, because the number of queues is by one smaller and thereby the virtual time is faster. Due to this, it could happen that also other F^k_{min} out of the second table become smaller than or equal to this new virtual time. If one of these is also the last packet in its queue, the virtual time will be even faster because the number of active queues becomes smaller again. This is called the iterated deletion problem. Therefore, we have to check Condition 7.20 again and go through the whole procedure, as long as there is any F^k_{min} smaller or equal to the virtual time in the second table.

If the packet leaving the system was the last packet in the last active queue, we re-initialize the system by resetting the virtual time and the queue number. The flowchart for the WFQ algorithm is given in Figure 7.11.

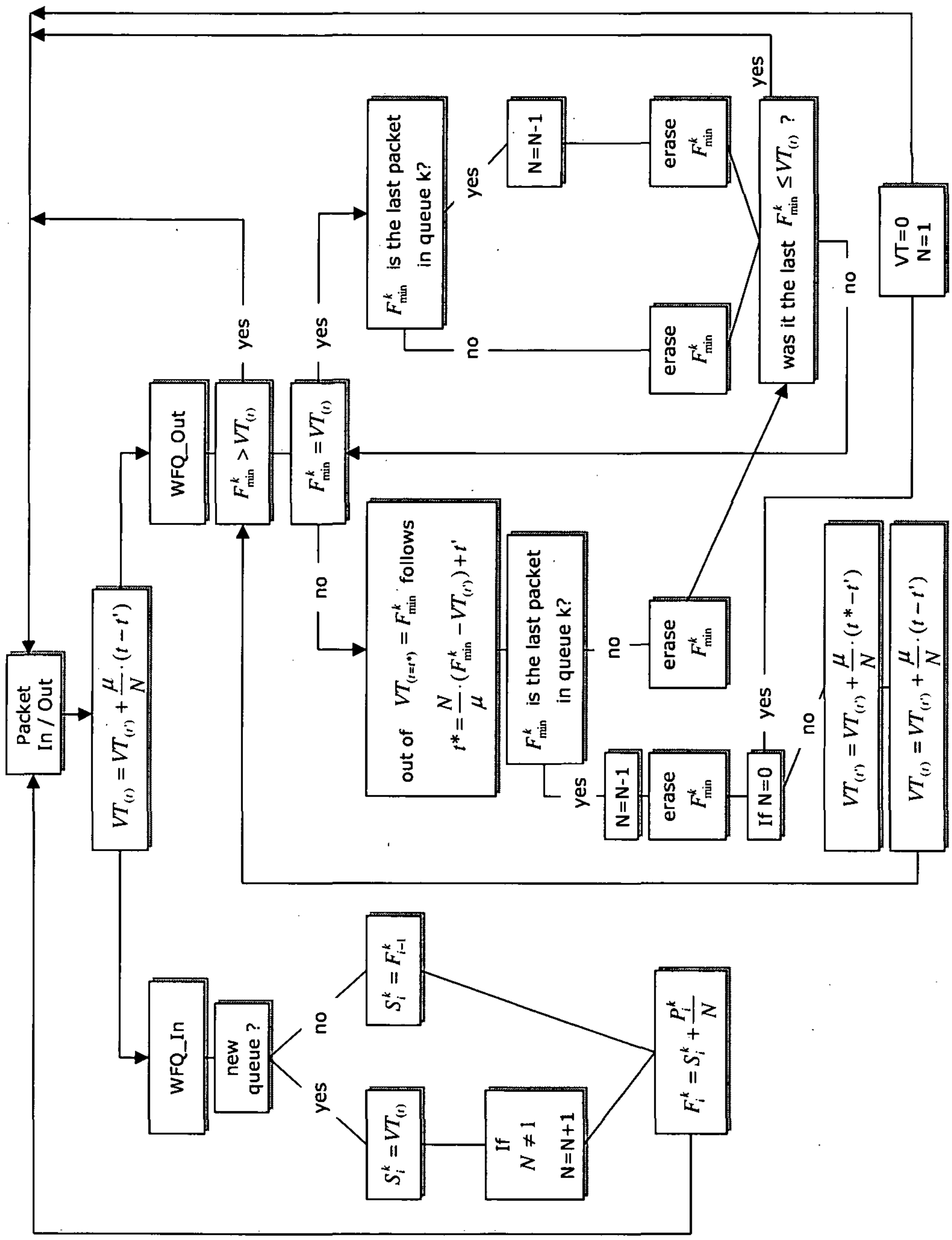

Figure 7.11: WFQ flowchart

To show how the WFQ simulation performs, we will take three queues – Q_1, Q_2, and Q_3. All three queues will generate the same amount of packets but with different size. The first queue (Q_1) will generate 50-bit packets only, the second queue (Q_2) 100-bit packets, and the third queue (Q_3) 150-bit packets. If the server is not overloaded, all packets out of the queues will be served (see Figure 7.12 for 95.1 and 99.03% server load). By decreasing the processing speed, the server load will be higher and therefore the processing of the packets from the queues will change.

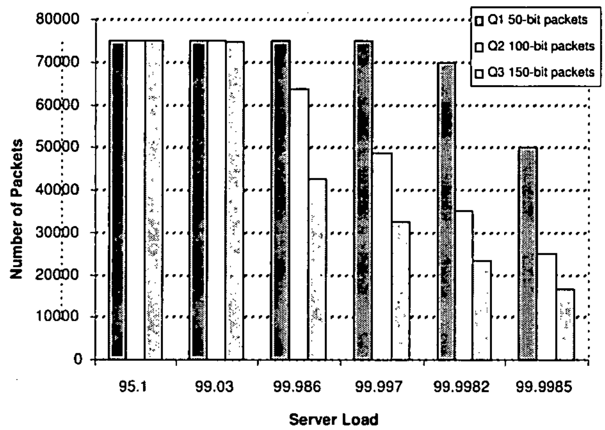

Figure 7.12: WFQ performance (packet-level)

The number of generated packets is a linear function of time. In this simulation, it is around 75.000 packets for each queue. The interesting part in the figures is where the server load is over 99%. In Figure 7.12, it can be seen that the amount of packets processed from each queue is different. Most packets are delivered from Q_1 followed by packets from Q_2, and finally Q_3. More significant is the information about the amount of bits processed, given by Figure 7.13. It can be seen that the amount of bits processed in the overload region is the same for each queue. This is exactly how WFQ should work. It would not be fair to handle the same amount of packets from each queue because the packets could be of different size.

The way how to favor one flow over the other is by setting different weights on each queue. We will use the same example as before, with Q_1, Q_2, and Q_3 and their corresponding deterministic packet sizes of 50, 100, and 150 bits.

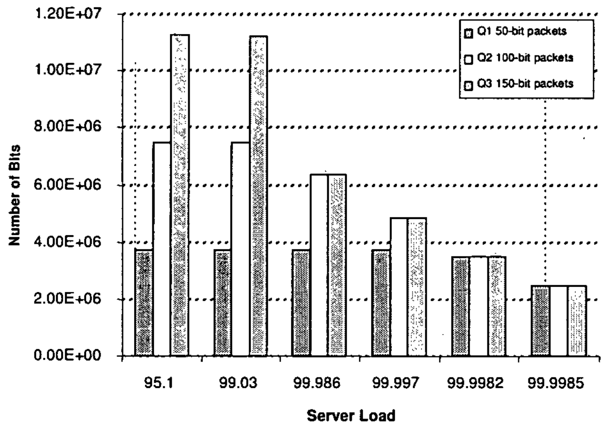

Figure 7.13: WFQ performance (bit-level)

Now we let the queuing system preferring Q_1, by giving Q_1 a weight twice as much as for Q_2 and Q_3. By this the weights are $W_1 = 2$, $W_2 = 1$, and $W_3 = 1$. Simulation results are shown in Figures 7.14 and 7.15.

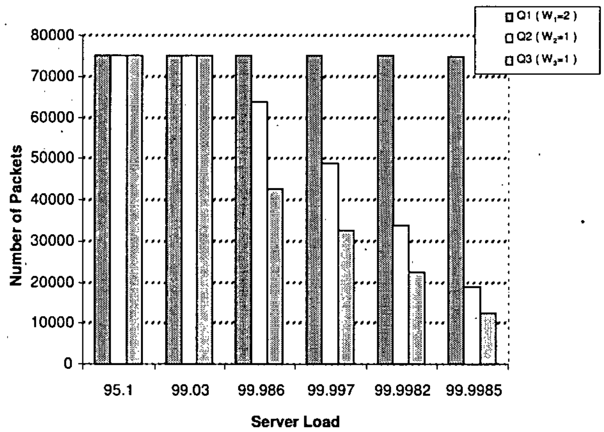

Figure 7.14: WFQ performance with weights (packet-level)

It can be seen that even in the region of higher load, queue Q_1 will now be preferred but at the price of decreased throughputs of Q_2 and Q_3. This is exactly what max-min fairness in Section 1.5.2 defines. In a fair system, no one can increase its share without decreasing the share of other users.

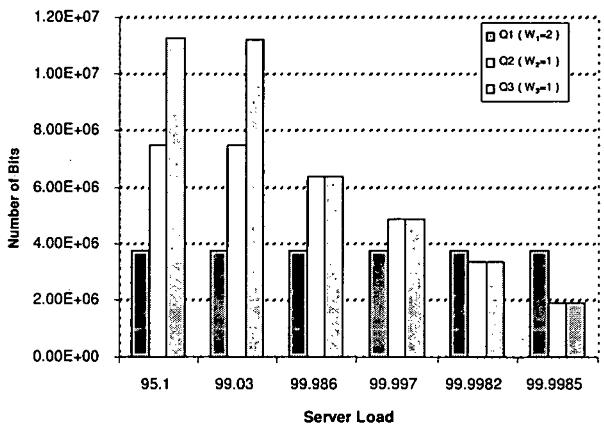

Figure 7.15: WFQ performance with weights (bit-level)

7.4 Simulation Results

We conducted two types of simulations by use of the models described in Section 7.2. The first is the packet delay comparison, and the second is the packet loss comparison.

7.4.1 Packet Delay Comparison

First, the QoS control case is simulated. Percentages of high priority packets are varied from 10% to 90%. The average packet delay for CBQ and WFQ are computed and plotted in Figures 7.16 and 7.17, respectively.

When using CBQ, as shown in Figure 7.16, the average packet delays of the high priority packets show lower values than those of low priority packets. This is due to the CBQ scheduling algorithm, i.e., only after the high priority queue is empty, low priority packets in the low priority queue are served. Each value in the curve is also plotted together with their corresponding standard deviation (2σ or 95%). Because the simulation run was long enough the values are relatively small. Therefore, the standard variation can almost not be seen in the graphical presentation. Numerical values are given in Table 7.3, where the standard deviation is given for high priority packets only.

When increasing the percentage of high priority packets, the average delay of

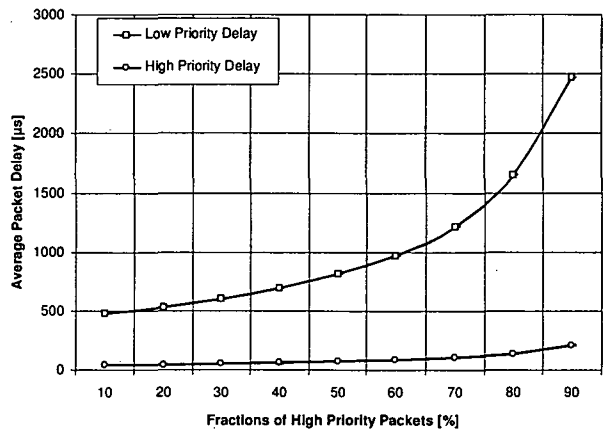

Figure 7.16: QoS control case with CBQ

Fraction of high priority packets [%]	10	20	30	40	50	60	70	80	90
High priority delay [μs]	40	45	50	57	68	82	102	137	208
Low priority delay [μs]	480	532	600	690	810	970	1213	1650	2473
Standard deviation (2σ or 95%)	0.98	0.82	0.89	1.09	1.32	1.58	2.66	3	3.2

Table 7.3: Numerical values of CBQ simulation

high priority packets itself also increases. When the amount of high priority packets increases, the server as well as line capacities take more time serving the high priority packets due to the fact that capacities are fixed. Although the amount of low priority packets decreases as the percentage of high priority packets increases, the average packet delay of low priority packets increases exponentially. Since increasingly more high priority packets are coming in, the server mostly serves high priority packets. Consequently, even though the amount of low priority packets is less, each low priority packet has to wait for a longer time to be served. This behavior of CBQ can lead to resource starvation for lower-priority traffic. Also, a misbehaving high-priority flow can contribute significantly to the amount of delay and jitter experienced by other high-priority flows sharing the same queue.

Being conscious of this drawbacks, we have choose CBQ because priority

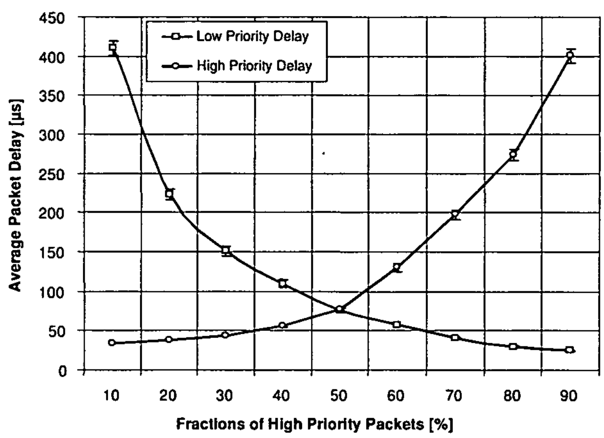

Figure 7.17:, QoS control case with WFQ

queuing disciplines places relatively low computational load on software-based routers, when compared with elaborate queuing disciplines such as WFQ. Additional CBQ allows routers to organize buffered packets and then service classes differently. For example, one can set priorities so that real-time applications, such as interactive voice and video, get priority over applications that do not operate in real time. It is also to the best advantage for signaling messages if they are transmitted over the same network as data. Fair queuing disciplines such as WFQ try to eliminate the starvation problem by classifying packets into flows and assigning the flows to a specifically dedicated queue. Queues are then served one packet at a time in round-robin order. In this model, the classifier splits the flows into only two queues - the high and the low priority one. If one of these two queues attempts to consume more than its fair share of bandwidth, then only this queue is affected, exactly what can be seen in Figure 7.17.

The relatively high difference in delay between the queues results out of the high input traffic amount that is over 90% of the link capacity. If traffic in one of the queues is high, the higher delay reflects the computational load of WFQ. Therefore, the sum of delays at the beginning and the end is higher than in the middle. WFQ enables also to put different weights on the queues, but for this part of simulation we put equal weights on the high and on the low priority queue. Numerical results are given in Table 7.4

Fraction of high priority packets [%]	10	20	30	40	50	60	70	80	90
High priority delay [μs]	34	38	44	56.1	77.5	130	197	273	400.5
Low priority delay [μs]	410.2	223	151	110	75.6	57	41	30	25
Standard deviation (2σ or 95%)	0.6	0.8	1.1	1.1	3	5	6.3	7	9.3

Table 7.4: Numerical values of WFQ simulation

In the next step, the link overprovisioning case is simulated. Figure 7.2 shows the simulation model and Figure 7.18 the computational results. In this case, it does not make any sense to order the incoming traffic into high and low priority queues nor to vary the amount of high priority packets, because in a FIFO system the delay will be the same for all packets. In this simulation, the variable is the transmission line speed, varying from 100 to 200 Mbit/s in steps of 10 Mbit/s.

As we can see in Figure 7.18, by increasing the transmission speed the average delay is dropping. In Table 7.5 numerical values for this case are given.

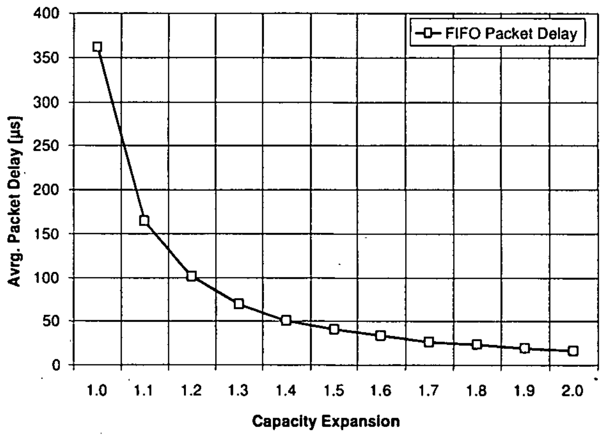

Figure 7.18: Overprovisioning

The last row comprises the server load. It is not surprising that delay and server load are dropping when transmission line speed increases. The results become interesting when comparing the curve with the results of the QoS controlled case. The main scope is the delay of high priority packets. In Figure 7.19 and

Server expansion	1.0	1.1	1.2	1.3	1.4	1.5	1.6	1.7	1.8	1.9	2.0
Packet delay [μs]	367	166	101	69.6	51	40.3	32.4	26.6	22.4	19.1	16.5
Server load [%]	90.4	82.2	75.3	69.5	64.6	60.3	56.5	53.1	50.2	47.6	45.2

Table 7.5: Numerical values of overprovisioning simulation

Figure 7.20, we added the values for the high priority packets from the QoS control simulation with CBQ and WFQ, respectively.

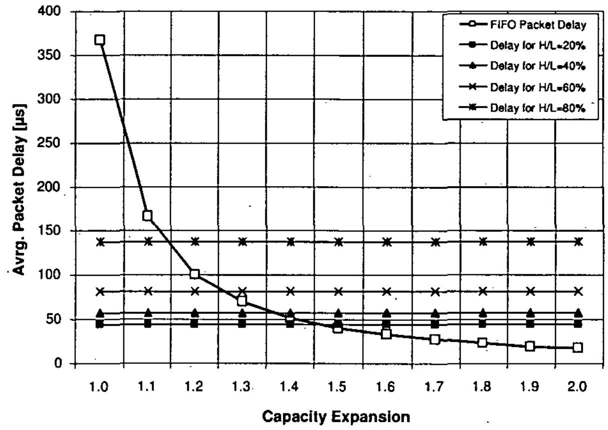

Figure 7.19: Comparison of FIFO delay and high priority delay (CBQ)

The values are plotted for the delay of high priority packets for the 20, 40, 60, and 80% fraction. It can be seen that for high fractions of high priority packets only a small increase of capacity in the overprovisioning model will lead to the same QoS as in the QoS control model. Even if the standard deviation is taken into account, it will not have a noticeable influence because of the low values (see Table 7.3). If the amount of high priority packets is very high, the QoS control model cannot prevent higher delay for high priority packets. We pointed this out in the beginning of this work - QoS mechanisms can be useful only if the network is not overloaded. To prevent overload, service level agreements or other access restriction mechanisms should be applied.

On the other hand, a fraction of 80% of high priority packets is rather a non-realistic assumption in a real network. By this, the more realistic fractions of high priority packets are the lower one, and this is exactly where it can be seen that, concerning packet delay, overprovisioning is not a good solution. From

Figure 7.19, we can see that providing the same delay for a 20% fraction of high priority packets, the overprovisioning model demands 47% more capacity. Considering also standard variation for the CBQ simulation within the 95% range, the resulting additional capacity would be between 46 and 48%.

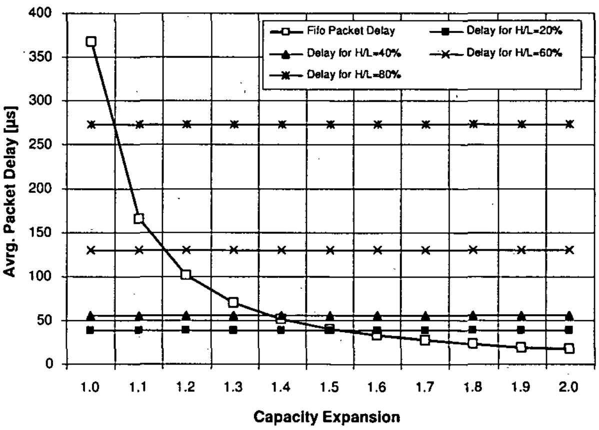

Figure 7.20: Comparison of FIFO delay and high priority delay (WFQ)

The same is true if comparing the results of the overprovisioning case with the QoS controlled simulation with WFQ (Figure 7.20). The only difference, if comparing it with CBQ queuing, is that with WFQ the amount of overprovisioning capacity is larger. For the 20% fraction of high priority, the amount of additional capacity would be between 48% and 52% if standard deviation is considered.

Generally, for a high fraction of high priority packets the link overprovisioning could be a good solution because minor increase in capacity would meet the need of high priority packets with QoS control. On the other hand, lowering the fraction of high priority packets leads to a decrease in packet delay in the QoS control model and this lower delay could be met only with much more bandwidth in the overprovisioning model.

7.4.2 Packet Loss Comparison

Next, a packet loss comparison is simulated. Here we use the same models as in the packet delay comparison, described in Figures 7.1 and 7.2. However, the buffer sizes of the queues are limited so that packet losses could be simulated. The total buffer size of each system is limited to ten times the average packet length, which is 750 bytes. By this, the buffer size is limited to 7.5 kbyte[1]. Other parameters remain the same as in Table 7.2. The simulation results for

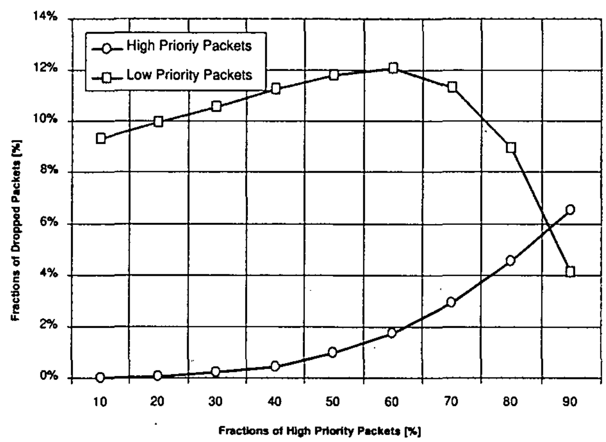

Figure 7.21: Packet loss - QoS control case with CBQ

the QoS control model with CBQ are shown in Figure 7.21.

As the total buffer size is limited to 7.5 kbyte, the high and the low priority queues obtain 3.75 kbyte each. The percentages of high priority packets are varied from 10% to 90%. Because the buffer sizes of the queues are limited, the results are quite different from the previous simulations. In the first part, until the 60% fraction of high priority packets, the number of dropped packets is increasing for both queues. As expected it is much higher for the low priority packets. If the fraction of high priority packets is more than 60% of the total amount of packets, the percentage of dropped low priority packets decreases. This is because high and low priority queues are treated separately and higher transmission speed is not enough in the case of incoming buffer overrun.

If comparing the previous results with the simulation results of dropping rate with WFQ scheduling, we will notice at first the fairness of WFQ (Figure 7.22). Both queues are handled only regarding the amount of traffic in the queue. If the amount is high, there is a high dropping rate. If it is lower,

[1]"k" is in place for 1000.

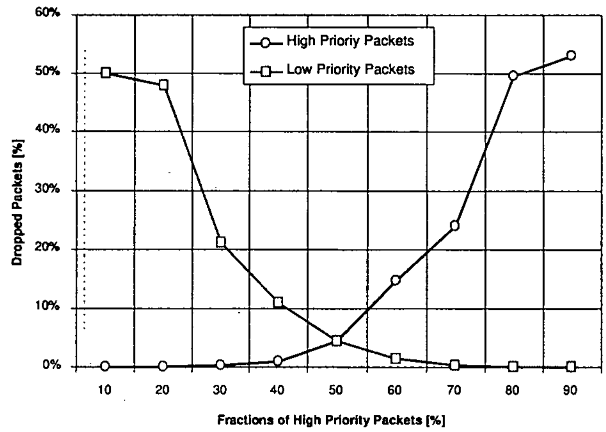

Figure 7.22: Packet loss - QoS control case with WFQ

the dropping rate is lower. Standard deviation is the same as in the case of delay simulations (+/- 1% for CBQ and +/- 2% for WFQ in the 95% range). In the case where the amount of high and low priority packets are the same (H/L=50%) the resulting dropping rate is 5% and this is the same value as the mean value for high and low priority dropping rate in the CBQ simulation curve (Figure 7.21).

In this simulation model, the buffer size of incoming queues is limited to 7.5 kbyte. It is interesting to notice that by increasing the size by 50%, the dropping rates are decreasing dramatically, especially in the high load region. With a buffer space of 7.5 kbyte, the dropping rates of the queues with a 90% fraction of total traffic, were about 50%. With 11.245 kbyte, this is decreasing to 30%. Also in the equilibrium state, it is falling from 5% to under 1%. The results are shown in Figure 7.23.

Next, the overprovisioning model is simulated. In this model, the limitation is done in the FIFO queue. The queue is limited to a buffer size of 7.5 kbyte. The simulation results have been depicted in Figure 7.24.

Because there is only one queue in the system, there is a common curve for the FIFO packets dropped. Increasing the transmission capacity of the system leads to a decrease of the amount of dropped packets. If the transmission capacity is twice the original, the amount of dropped packets reaches zero.

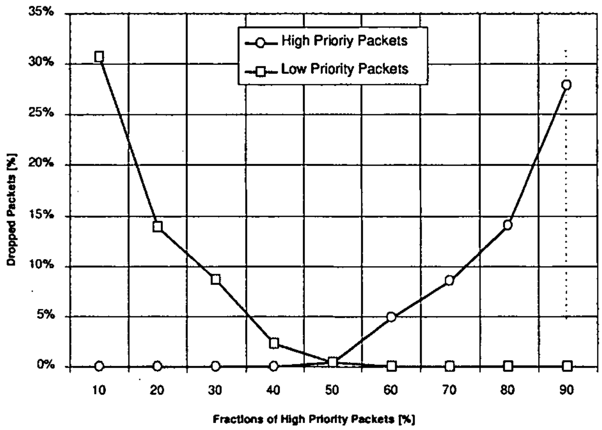

Figure 7.23: Packet loss - QoS control case with WFQ and higher buffer space

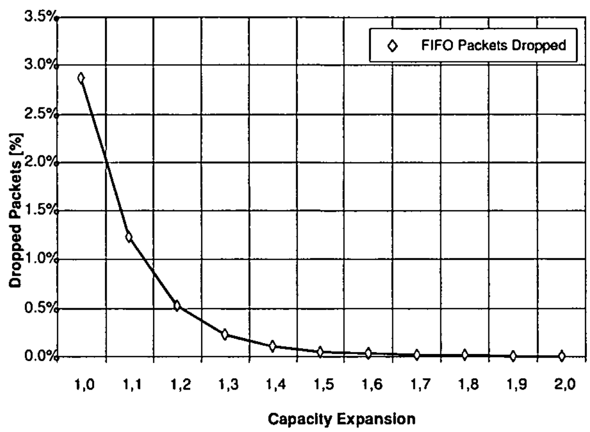

Figure 7.24: Packet loss - overprovisioning model

Similarly as in the previous simulations, we will compare these results with the values for the dropping rate of the high priority packets of the QoS control system. As we can see in Figure 7.25, the results are the same as in the simulation for packet delay. The difference between CBQ and WFQ is only a smaller dropping rate for the higher priority packets in CBQ. Low fractions of high priority packets need high amount of capacity overprovisioning and vice versa.

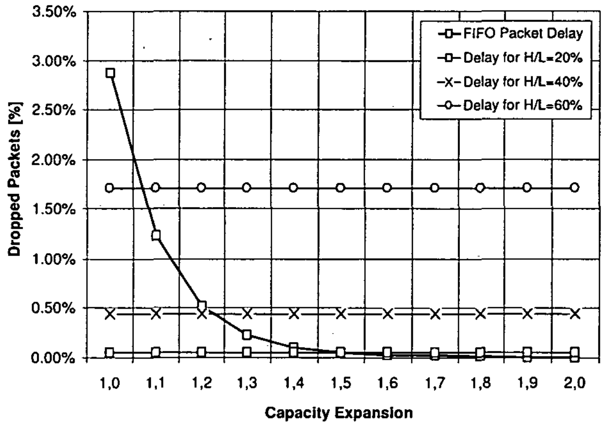

Figure 7.25: Comparison of FIFO dropping and high priority dropping

7.4.3 Cascaded QoS Control Simulation Model

By cascading the QoS control simulation model, we are able to compare WFQ and CBQ on a cascaded basis. The QoS control simulation model for one QoS stage is once again depicted in Figure 7.26.

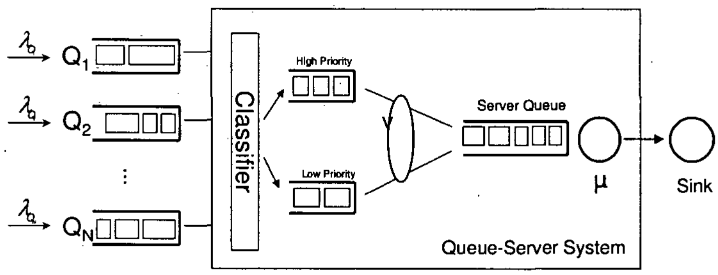

Figure 7.26: Single QoS control simulation model

In the previous simulations, the output of the queue-server system terminated directly into the sink. In this part, the output of the first system will provide the input for the second one. Altogether, there are four identical queue-server systems. Figure 7.27 shows the cascaded model. The simulation parameters for each stage are the same as before (see Table 7.2).

The delay time for the packets will now be the sum of the delays from each

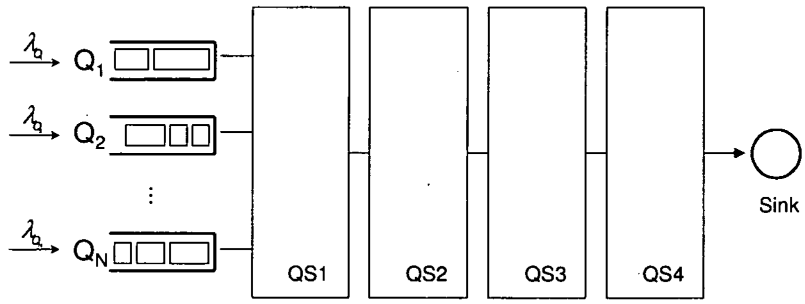

Figure 7.27: Cascaded QoS control simulation model

of the four queue-server systems. The simulation results indicated that the delay time (except for the first stage) is nearly uniformly distributed over each system, because each system classifies and processes the packets in the same manner. This means the first classification stage is dominant. At this point the advantage of MPLS should be pointed out. MPLS make the classification only at the first stage. Once labelled at the ingress router, the packets do not have to be classified in the core any more, resulting in a smaller overall delay.

It turned out that the mean delay of the packets arriving at the sink is approximately four times the delay of the single queuing model. Results for CBQ and WFQ scheduling are given in Figure 7.28 and Figure 7.29, respectively.

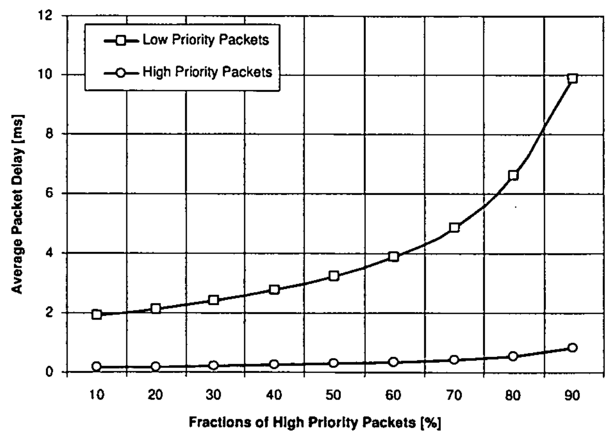

Figure 7.28: Cascaded QoS control with CBQ

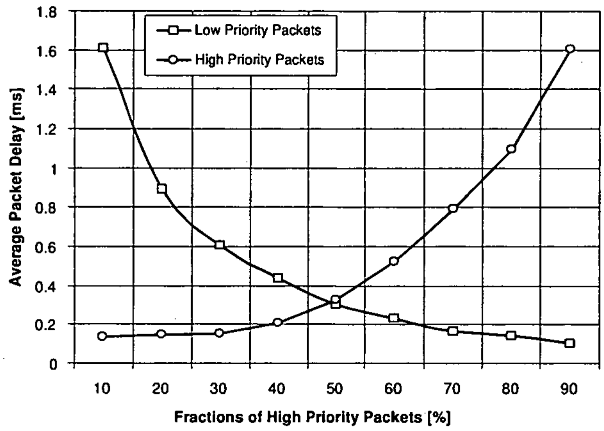

Figure 7.29: Cascaded QoS control with WFQ

Table 7.6 shows the numerical values of the variance of the given cascaded delays for CBQ and WFQ scheduling, respectively. Zero Variance would mean the same delay for all packets arriving at the sink, thus indicating no jitter at all. In CBQ, the high priority packets have a smaller variance than the low priority packets. By processing high priority packets as soon as they arrive in the system, jitter has been kept low. The variance of the delay time for low priority packets is much higher, specially if the fraction of high priority packets is high.

Fraction of high priority packets [%]	CBQ		WFQ	
	High priority	Low priority	High priority	Low priority
10	1.12e-9	2.41e-7	2.43e-5	2.28e-5
20	1.49e-9	3.04e-7	6.09e-6	6.70e-6
30	2.01e-9	3.96e-7	2.55e-6	3.46e-6
40	2.77e-9	5.36e-7	1.05e-6	1.93e-6
50	4.04e-9	7.61e-7	9.89e-7	9.50e-7
60	6.13e-9	1.15e-6	1.87-6	1.16e-6
70	1.02e-8	1.94e-6	3.60e-6	2.87e-6
80	1.91e-8	3.90e-6	5.05e-6	4.86e-6
90	4.53e-8	1.07e-5	1.81e-5	2.03e-5

Table 7.6: Variance of the delay of cascaded QoS control

For WFQ, the variance of the mean delay for high and low priority packets is almost equal. It only depends whether the queues are equally filled, or one queue is much larger than the other queue. Larger differences in queue length reflect higher variances for both queues. Equalizing the variance for all queues is one of the advantages of WFQ.

For WFQ, Figure 7.30 and Figure 7.31 show the graphical presentation of the variance of the delay time for high and low priority packets, respectively.

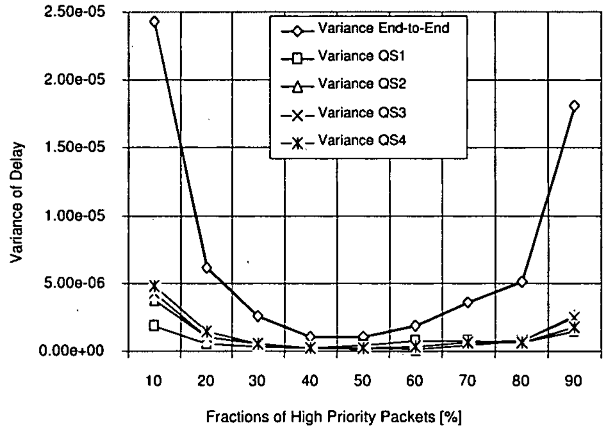

Figure 7.30: Variance of delay for high priority packets with WFQ

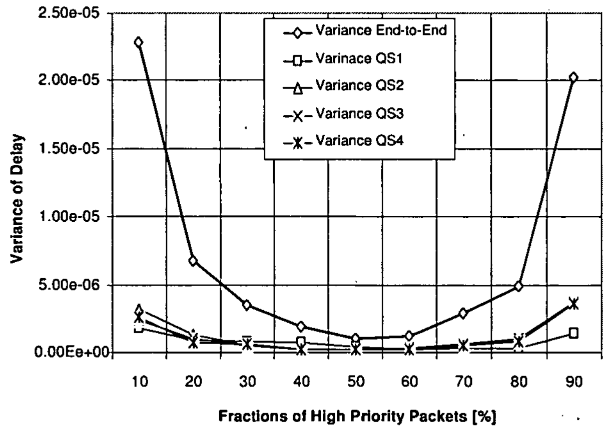

Figure 7.31: Variance of delay for low priority packets with WFQ

The figures also include the variance of the delay at each of the four queue-server systems.

Figure 7.32 and Figure 7.33 present the variance of delay for CBQ case.

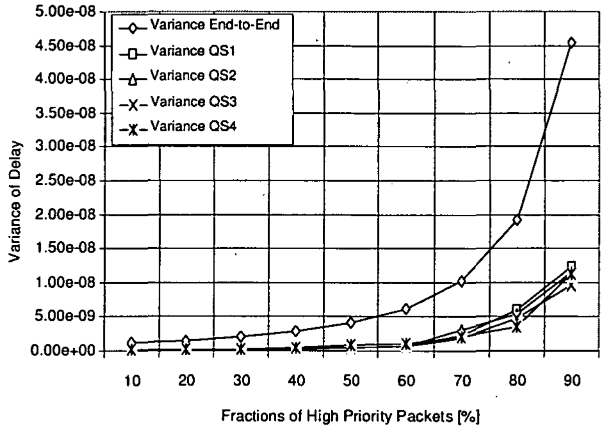

Figure 7.32: Variance of delay for high priority packets with CBQ

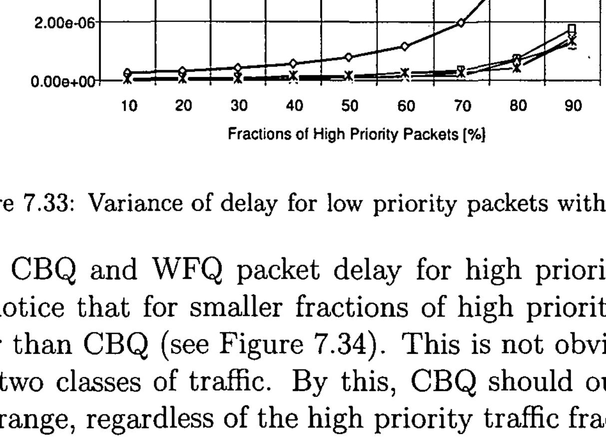

Figure 7.33: Variance of delay for low priority packets with CBQ

Comparing the CBQ and WFQ packet delay for high priority packets, it is interesting to notice that for smaller fractions of high priority packets WFQ performs better than CBQ (see Figure 7.34). This is not obvious, because we are using only two classes of traffic. By this, CBQ should outperform WFQ over the whole range, regardless of the high priority traffic fraction. The more classes are defined, the better WFQ will perform compared to CBQ. Also for the highest priority class in CBQ.

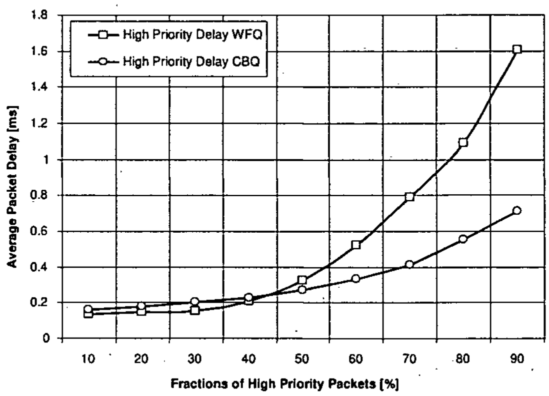

Figure 7.34: High packet delay with CBQ and WFQ

For low priority traffic, WFQ is always better than CBQ (see Figure 7.35). Especially in the region where the fraction of high priority traffic is higher.

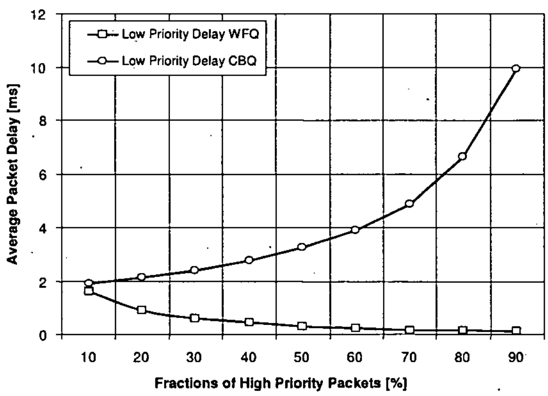

Figure 7.35: Low packet delay with CBQ and WFQ

According to the results of this section, the conclusion is that CBQ performs better only for high priority packets, if the fraction of high priority packets is over 50 % of the overall amount of traffic. This affects the mean delay time as well as the variance of the delay. All other performance values are more in support of WFQ. If high priority packets would have a higher weight than low priority packets, WFQ would clearly outperform CBQ over the whole range of the simulation results.

8 Conclusion and Future Work

This work has concentrated on the network part of the third generation mobile network, UMTS. In Release 6 of UMTS specifications, the whole network will be based on packet switching. In Chapter 5, a possible QoS architecture was shown, where the Differentiated Service architecture of the IETF is adapted by defining two different DiffServ domains - the access part and the network part. They are interconnected to each other by border gateways. The realization is simple and pushes the complexity to the edge of the network.

The predominant technology used in the core will be WDM-based and by this overprovisioning could be a QoS alternative. This could really be put into effect only if the amount of traffic is not forcing the network to operate at full capacity. On the other hand, if there is a lot of spare capacity, there is no demand for QoS mechanisms because there is no competition for network resources between applications. By this, only the state of sustained network overload, where demand for network load is close to the capacity, demands for QoS mechanisms.

The simulation results in this last chapter clearly showed that if we have a balanced amount of high and low priority packets in the network, overprovisioning should be the choice of dealing with higher delays and dropping rates. The truth is that having a balanced relation or even overweight number of high priority packets in the network is not a realistic assumption. Especially not in the fact that Location-Based Services will produce a lot of traffic but only a part of this will be time sensitive like speech transmission. Moreover, LBS applications will not diminish the importance of speech transmission or even make it obsolete. It should be kept in mind that voice transmission will still be of vital interest to every mobile network supplier. Therefore, we have to find ways of protecting real-time traffic from non-real-time traffic. On the other hand, it is realistic to expect that the amount of traffic produced by voice transmission will only be a small part of the whole picture. This leads to further results of the simulations - small amount of packets with high priority demand a lot of overprovisioning to achieve the same QoS performance, if compared to simple QoS mechanisms like implementation of scheduling mechanisms such as Class Based Queuing with strict priority. There are even more complex mechanisms like Weighted Fair Queuing that are even better, for the price of a more complex algorithm that has to be implemented in the router.

Evaluating implementation cost of end-to-end QoS mechanisms, but also the influence of end-system based QoS like forward error correction, rate control and packetization will be part of future work. In this context, also Service Level Agreements will not be let out of the scope. In order to take advantage of the results, future investigations will also concentrate on practical deployment and operational experience.

Appendix A Position Calculation Function

The location estimate is performed by a Position Calculation Function (PCF) located in the network or, if the mobile station has the capability, in the mobile station itself. With the same network architecture, mobile station functions, LMU functions and measurement inputs, the PCF can be based on one of two possible variants of E-OTD, known as "circular" and "hyperbolic"[ETSI01].

A.1 Circular Variant

The E-OTD Circular variant (E-OTD-C) measures the arrival time from each of the base stations, individually at the mobile station and LMU.

There are five quantities associated with E-OTD-C:

- Observed Time at the mobile station (**MOT**) at which a signal arrives from a base station. This is a time measured against the mobile station's internal clock.
- Observed Time at the LMU (**LOT**) at which a signal arrives from a base station. This is a time measured against the LMU's internal clock.
- Time offset (ε) between the mobile station's internal clock and the LMU's internal clock.
- The geometrical distance from mobile station to base station (**DMB**).
- The geometrical distance from LMU to base station (**DLB**).

These quantities are related by

$$DMB - DLB = \nu(MOT - LOT + \varepsilon) \tag{A.1}$$

in which ν is the speed of the radio waves. There will be one such equation for each base station. Since there are three unknown quantities (mobile station position x, y and clock offset ε) measurements to at least three base stations are required to solve for the mobile station location and the unknown clock offset ε. The estimate of the position of the mobile station is defined by the intersection of circles centred on the base stations common to observations made by the mobile station and LMUs.

The uncertainty associated with the radius of a circle is known as the measurement error margin. The overlap of the resulting areas is defined by a confidence ellipse, described by the length of its axes and its orientation.

A.2 Hyperbolic Variant

The hyperbolic and circular methods differ in the relationship between the mobile station measurement error margin and the geographic location of the mobile station relative to base stations. In all other respects the implementation is identical.

Appendix B Abbreviations

3GPP	3RD GENERATION PARTNERSHIP PROJECT
3GPP2	3RD GENERATION PARTNERSHIP PROJECT 2
A-GPS	ASSISTED GLOBAL POSITIONING SYSTEM
ADSL	ASYMMETRICAL DIGITAL SUBSCRIBER LINE
AF	ASSURED FORWARDING
AFLT	ADVANCED FORWARD LINK TRILATERATION
AME	ADVANCED MAXIMUM LIKELIHOOD ESTIMATION
AOA	ANGLE OF ARRIVAL
API	APPLICATION INTERFACE
ARIS	AGGREGATE ROUTE-BASED IP-SWITCHING
ARPANET	ADVANCED RESEARCH PROJECTS AGENCY NETWORK
AT&T	AMERICAN TELEPHONE AND TELEGRAPH
ATM	ASYNCHRONOUS TRANSFER MODE
BBN	BOLT, BERANEK AND NEWMANN
BCCH	BROADCAST COMMON CONTROL CHANNEL
BG	BORDER GATEWAY
BS	BASE STATION
BSC	BASE STATION CONTROLLER
BSS	BASE STATION SUBSYSTEM
BTS	BASE TRANSCEIVER STATION
CBC	CELL BROADCAST CENTER
CBQ	CLASS BASED QUEUING
CDMA	CODE DIVISION MULTIPLE ACCESS
CE	CONGESTION EXPERIENCE

CI	Cell ID
CLP	Cell Loss Priority
COO	Cell of Origin
CORBA	Common Object Request Broker Architecture
COS	Class-of-Service
CPICH	Common Pilot Channel
CSFQ	Core Stateless Fair Queuing
CU	Currently Unused
D-GPS	Differential Global Positioning System
DCM	Database Correlation Method
DEGA	Defined Geographical Areas
Delay-EDD	Delay Earliest Due Date
DHCP	Dynamic Host Configuration Protocol
DiffServ	Differentiated Services
DoD	Department of Defence
DRR	Deficit Round Robin
DS	Differentiated Services
DSCP	Differentiated Services Code Point
DTV	Digital Television
DWDM	Dense Wavelength Division Multiplexing
E-OTD	Enhanced Observed Time Difference
E911	Enhanced 911
EC	European Commission
ECN	Explicit Congestion Notification
EDD	Earliest Due Date
EDD-FB	Earliest Due Date for Finite Buffer
EDF	Earliest Deadline First
EDGE	Enhanced Data Rates for GSM Evolution
EF	Expedited Forwarding
EGNOS	European Geostationary Navigation Overlay Service
ESA	European Space Agency
ETSI	European Telecommunication Standards Institute
ExD	Expected Deadline
FCC	Federal Communication Commission

FCFS	First Come First Serve
FDD	Frequency Division Duplex
FEC	Forward Equivalence Class
FIFO	First-in First-out
FQ	Fair Queuing
FR	Frame Relay
FTP	File Transfer Protocol
GCIM	Ground Control Integrity Monitoring
GDP	Geometric Dilution of Position
GERAN	GSM EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GIS	Geographic Information Systems
GLONASS	Global Navigation Satellite System
GMLC	Gateway Mobile Location Center
GPRS	General Packet Radio Service
GPS	Generalized Processor Sharing
GPS	Global Positioning System
GSM	Global System for Mobile Communication
GSSGP	Base Station System GPRS Protocol
GTD	Geometrical Time Difference
GTP	GPRS Tunnelling Protocol
HLR	Home Location Register
HOL-PS	Head of Line Processor Sharing
HRR	Hierarchical Round Robin
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
I-mode	Information mode
ICMP	Internet Control Message Protocol
ID	Identification
IESG	Internet Engineering Steering Group
IETF	Internet Engineering Task Force
IIOP	Internet Inter-ORB Protocol
IN	Intelligente Network
IntServ	Integrated Services

IP	Internet Protocol
IPDL	Idle Period Downlink
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISSLL	Integrated Services over Specific Link Layer
IT	Information Technology
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union, Telecommunications Standardization Section
ITU-R	International Telecommunication Union, Radiocommunications Standardization Section
Jitter-EDD	Jitter Earliest Due Date
LA	Location Area
LAC	Local Area Code
LAMGA	Location of All Mobiles in Geographical Area
LAP	Location Aided Network Planning
LBAP	Linear Bounded Arrival Process
LBS	Location-Based Services
LC	Location Class
LCF	LCS Client Function
LCS	Location Services
LDP	Label Distribution Protocol
LER	Label Edge Routing
LIF	Location Interoperability Forum
LMU	Location Measurement Unit
LocDC	Location Drafting Committee
Loran	Long Range Navigation
LOS	Line-of-Sight
LSCF	LCS Server Control Function
LSOF	LCS System Operation Function
LSP	Label Switched Path
LSR	Label Switched Router
MAHO	Mobile Assisted Handoff
MAP	Mobile Application Protocol

MBG	Mobile Border Gateway
MCC	Mobile Network Code
MF	Multi Field
MGIF	Mobile Gaming Interoperability Forum
MLC	Mobile Location Center
MLP	Mobile Location Protocol
MMS	Multimedia Massaging
MMS-IOP	MMS Interoperability Group
MNC	Mobile Network Code
MoU	Memorandum of Understanding
MP3	MPEG 1 Layer-3
MPEG	Motion Picture Experts Group
MPLS	Multi-Protocol Label Switching
MS	Mobile Station
MSISDN	Mobile Station ISDN Number
MT	Mobile Terminal
MWIF	Mobile Wireless Internet Forum
NCSA	National Center for Supercomputing Application
NELS	North West European Loran-C System
NGI	Next Generation Internet
NGN	Next Generation Network
NLOS	Non Line-of-Sight
NMR	Network Measurement Results
NSIS	Next Steps in Signaling
NSS	Network Subsystem
NTT	Nippon Telephone and Telegraph
O&M	Operation and Maintenance
OECD	Organisation for Economic Cooperation and Development
OGC	Open GIS Consortium
OGC	Open Group Consortium
OMA	Open Mobile Alliance
OpenLS	Open GIS Location Services
ORB	Object Request Broker
OSA	Open Service Access
OTD	Observed Time Difference

OTDA	Observed Time Difference of Arrival
PCF	Position Calculation Function
PCS	Personal Communication Services
PDA	Personal Digital Assistant
PDE	Positioning Determination Equipment
PDP	Packet Data Protocol
PGPS	Packet by Packet General Processor Sharing
PHB	Per-Hop Behavior
PLMN	Public Land Mobile Networks
PQ	Priority Queuing
PRCF	Position Radio Coordination Function
PRN	Pseudo Random Noise
PRRM	Position Radio Resource Management
PS	Processor Sharing
PSAP	Public Safety Answering Points
PSMF	Position System Measurement Function
PSTN	Public Switched Telephone Network
PVC	Permanent Virtual Circuit
QoS	Quality-of-Service
RAB	Radio Access Bearer
RED	Random Early Detection
RF	Radio Frequency
RFC	Request for Comment
RNC	Radio Network Controller
RRLP	Radio Resource LCS Protocol
RSVP	Resource Reservation Protocol
RTD	Real Time Difference
RTT	Round Trip Time
RXLEV	Receiving Power Level
SA	Selective Availability
SAI	Service Area Identifier
SCEDD	Service Curve based Earliest Due Date
SCFQ	Self Clocked Fair Queuing

SDH	Synchronous Digital Hierarchy
SFN	System Frame Number
SFQ	Stochastic Fairness Queuing
SGSN	Serving GPRS Support Node
SIG	Special Interest Group
SIM	Subscriber Identity Module
SLA	Service Level Agreement
SLoP	Spatial Location Protocol
SMLC	Serving Mobile Location Center
SMR	Specialized Mobile Radio
SMS	Short Message Service
STE	Shortest Time to Extinction
STFQ	Start Time Fair Queuing
STK	SIM Application Toolkit
STM	Synchronous Transfer Mode
TA	Timing Advance
TA-IPDL	Time Aligned Idle Period Downlink
TCA	Traffic Conditioning Agreement
TCP	Transmission Control Protocol
TCS	Traffic Conditioning Specification
TDD	Time Division Duplex
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TE	Terminal Equipment
TOE	Time of Emission control mode
TOS	Type of Service
TS	Time Stamp
TTL	Time-to-Live
UDP	User Datagram Protocol
UE	User Equipment
UHF	Ultra High Frequency
UL-TOA	Uplink Time-of-Arrival
UMTS	Universal Mobile Telecommunication System
UTRA	UMTS Terrestrial Radio Access
UTRA-FDD	UMTS Terrestrial Radio Access/

	FREQUENCY DIVISION DUPLEX
UTRA-TDD	UMTS TERRESTRIAL RADIO ACCESS/ TIME DIVISION DUPLEX
UWB	ULTRA WIDEBAND
VAS	VALUE ADDED SERVICES
VC	VIRTUAL CLOCK
VCI	VIRTUAL CIRCUIT IDENTIFIER
VLR	VISITOR LOCATION REGISTER
VMSC	VISITED MOBILE SWITCHED CENTER
VoIP	VOICE OVER INTERNET PROTOCOL
VPI	VIRTUAL PATH IDENTIFIER
VPN	VIRTUAL PRIVATE NETWORK
VT	VIRTUAL TIME
W^2FQ	WORST CASE WFQ
WF^2Q	WORST CASE FAIR WEIGHTED FAIR QUEUING
W3C	WWW CONSORTIUM
WAD	WIDE AREA DIFFERENTIAL
WAN	WIDE AREA NETWORK
WAP	WIRELESS APPLICATION PROTOCOL
WDM	WAVELENGTH DIVISION MULTIPLEXING
WFQ	WEIGHTED FAIR QUEUING
WGS	WORLD GEODETIC SYSTEM
WiMax	WORLDWIDE INTEROPERABILITY FOR MICROWAVE ACCESS
WLAN	WIRELESS LOCAL AREA NETWORK
WRED	WEIGHTED RANDOM EARLY DETECTION
WRR	WEIGHTED ROUND ROBIN
WTO	WORLD TRADE ORGANISATION
WWW	WORLD WIDE WEB
XML	EXTENSIBLE MARKUP LANGUAGE

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