CHAPTER 1

INTRODUCTION

Wireless communications is an emerging field, which has seen enormous growth in the last several years. The huge uptake rate of mobile phone technology, Wireless Personal Area Networks (WPANs) and the exponential growth of Internet has resulted in an increased demand for new methods of obtaining high capacity wireless networks.

The Fourth generation (4G) wireless networks are expected to provide high data rate transmission of multimedia services such as wireless video, wireless internet access and mobile computing. The two most promising wireless networks are cellular networks and Wireless Local area Networks (WLANs). The second generation (2G) cellular networks such as GSM and IS 95 were a revolution from analog to digital technology. 2.5 G cellular networks such as General Packet Radio Services (GPRS) provide packet switched low rate typically up to 100 kbps for data services. The evolution from second to third generation mobile networks, wideband radio access and internet based protocols pave the way from mobile handset today to a mobile multimedia device in future.

The rapid expansion of the mobile market over the past few years has seen cellular communications move away from just voice to a host of multimedia services. Users are now demanding their handsets to be packed with more features while at the same time being lighter and more power efficient. As a result, third generation (3G) Wideband Code Division Multiple
Access (WCDMA) mobile communications are gearing up to deliver the kind of flexible services wanted.

More research has been conducted in this field by research groups around the world. CDMA supports variable bit rates and hence is the ideal mode of communication for future cellular networks. To support various integrated services with a certain quality of service (QoS) requirement in these wireless networks, resource provisioning is a major issue (Grillo et al 1998, Hong and Rappaport 1986). The Universal Mobile Telecommunication Systems (UMTS) supports QoS provisioning through four basic classes of service.

- Class 1: Conversational (high sensitivity to delay and jitter).
- Class 2: Streaming (medium sensitivity to delay and high sensitivity to jitter).
- Class 3: Interactive (low sensitivity to delay, high sensitivity to round trip delay time and Bit Error Rate (BER)).
- Class 4: Background (no delay sensitivity, high sensitivity to BER).

Call admission control (CAC) is a QoS provisioning strategy to limit the number of connections into the networks in order to reduce the network congestion and call dropping. In previous generation networks such as AMPS, GSM, GPRS, the decision of accepting a new call was a relatively easy one, since the available number of channels in a cell is known. CDMA on the other hand is interference limited and the number of calls cannot specify the capacity of the system. A user will be granted access to the network only if this connection will not cause the other users to experience a drop in quality or affect system stability. In wireless networks, call dropping
is also possible due to the mobility of the users. Since mobile users may change cells a number of times during the lifetime of their connections, availability of wireless network resources at the connection setup time does not necessarily guarantee that wireless network resources are available throughout the lifetime of a connection. Thus users may experience performance degradations due to mobile handoffs. This problem will be magnified in future micro/pico-cellular networks, where handoff events may occur at a much higher rate compared to today’s macro-cellular systems. Call admission control (CAC) and bandwidth reservation are required to address this problem. Since forced call terminations due to handoff blocking are generally more objectionable than new call blocking, the probability of handoff dropping ($P_{hd}$) is considered as the key connection-level QoS metric provisioned by CAC in wireless cellular networks. As it is impractical to completely eliminate handoff call dropping, the best one could do is to keep $P_{hd}$ below a target level. Moreover, maximizing resource utilization while keeping the probability of new call blocking ($P_{nb}$), below a target value is another critical factor for evaluating the performance of CAC algorithms. A good CAC scheme has to balance call blocking and call dropping in order to provide the desired QoS requirements.

The main objectives of this thesis are:

- Survey of various Call Admission Control, Resource Allocation and reservation schemes in WCDMA networks.
- Survey some of the improvements to Call Admission and Resource reservation mechanisms proposed in the literature.
- Introduce and develop improvements to the existing schemes to further reduce call dropping rates and further prioritize services according to user requests.
• Investigate through simulations the performance achieved by the suggested improvements.

1.1 MOBILE COMMUNICATION NETWORKS AND RESOURCE MANAGEMENT ISSUES

Mobile communication networks have developed at an astounding speed during the past few decades. The cellular concept arose from the need to share the spectrum which is a limited resource in mobile communication. Before discussing the actual content of this thesis, it is useful to understand the contextual background of the subject.

1.1.1 First Generation Analog Systems

In 1980 the mobile cellular era had started, and since then mobile communications have undergone significant changes and experienced enormous growth. Figure 1.1 shows the evolution of the mobile networks.

First-generation mobile systems used analog transmission for speech services. In 1979, the first cellular system in the world became operational by Nippon Telephone and Telegraph (NTT) in Tokyo, Japan. The system utilised 600 duplex channels over a spectrum of 30 MHz in the 800 MHz band, with a channel separation of 25 kHz. Two years later, the cellular epoch reached Europe. The two most popular analog systems were Nordic Mobile Telephones (NMT) and Total Access Communication Systems (TACS). In 1981, the NMT-450 system was commercialised by NMT in Scandinavia. The system operated in the 450 MHz and 900 MHz band with a total bandwidth of 10 MHz. TACS, launched in the United Kingdom in 1982, operated at 900 MHz with a band of 25 MHz for each path and a channel bandwidth of 25 kHz. Extended TACS was deployed in 1985. Other than
NMT and TACS, some other analog systems were also introduced in 1980s across the Europe. For example, in Germany, the C-450 cellular system, operating at 450 MHz and 900 MHz (later), was deployed in September in 1985. All of these systems offered handover and roaming capabilities but the cellular networks were unable to interoperate between countries. This was one of the inevitable disadvantages of first-generation mobile networks. In the United States, the Advanced Mobile Phone System (AMPS) was launched in 1982. The system was allocated a 40-MHz bandwidth within the 800 to 900 MHz frequency range. In 1988, an additional 10 MHz bandwidth, called Expanded Spectrum (ES) was allocated to AMPS.

Figure 1.1 Evolution of mobile networks (Holma and Toskala 2002).

1.1.2 Second-Generation Mobile Systems

Second-generation (2G) mobile systems were introduced in the end of 1980s. Low bit rate data services were supported as well as the traditional speech service. Digital transmission rather than analog transmission was used by these systems. Consequently, compared with first-generation systems, higher spectrum efficiency, better data services, and more advanced roaming
were offered by 2G systems. In Europe, the Global System for Mobile Communications (GSM) was deployed to provide a single unified standard. This enabled seamless services throughout Europe by means of international roaming. The earliest GSM system operated in the 900 MHz frequency band with a total bandwidth of 50 MHz. During the 20 year period of development, GSM technology has been continuously improved upon to offer better services in the market. New technologies have been developed based on the original GSM system, leading to some more advanced systems known as 2.5 Generation (2.5G) systems. Until now as far as the largest mobile system worldwide is concerned, GSM is the technology of choice in over 190 countries with about 787 million subscribers (GSMweb).

In the United States, there were three lines of development in second-generation digital cellular systems. The first digital system, introduced in 1991, was the IS-54 (North America TDMA Digital Cellular), of which a new version supporting additional services (IS-136) was introduced in 1996. Meanwhile, IS-95 (cdma One) was deployed in 1993. The US Federal Communications Commission (FCC) also auctioned a new block of spectrum in the 1900 MHz band (PCS), allowing GSM1900 to enter the US market. In Japan, the Personal Digital Cellular (PDC) system, originally known as JDC (Japanese Digital Cellular) was initially defined in 1990. Commercial service was started by NTT in 1993 in the 800 MHz band and in 1994 in the 1.5 GHz band. Table 1.1 shows the technical parameters of four typical second generation digital mobile systems.

Nowadays, second-generation digital cellular systems still dominate the mobile industry throughout the world. However, they are evolving towards third generation (3G) systems because of the demands imposed by increasing mobile traffic and the emergence of new type of services. The new systems, such as HSCSD (High Speed Circuit Switched Data), GPRS
Table 1.1 Technical parameters of second-generation digital systems

<table>
<thead>
<tr>
<th></th>
<th>GSM</th>
<th>IS-136</th>
<th>IS-95</th>
<th>PDC</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Multiple access</strong></td>
<td>TDMA</td>
<td>TDMA</td>
<td>CDMA</td>
<td>TDMA</td>
</tr>
<tr>
<td><strong>Modulation</strong></td>
<td>GMSK</td>
<td>p/4-DQPSK</td>
<td>Coherent p/4-DQPSK</td>
<td>QPSK/O-QPSK</td>
</tr>
<tr>
<td><strong>Carrier spacing</strong></td>
<td>200 kHz</td>
<td>30 kHz</td>
<td>1.25 MHz</td>
<td>25 kHz</td>
</tr>
<tr>
<td><strong>Carrier bit rate</strong></td>
<td>270.833 kbit/s</td>
<td>48.6 kbit/s (p/4-DQPSK)</td>
<td>72.9 kbit/s (8-PSK)</td>
<td>1.2288 Mchip/s</td>
</tr>
<tr>
<td><strong>Frame Length</strong></td>
<td>4.615 ms</td>
<td>40 ms</td>
<td>20 ms</td>
<td>20 ms</td>
</tr>
<tr>
<td><strong>Slots per frame</strong></td>
<td>8/16</td>
<td>6</td>
<td>1</td>
<td>3/6</td>
</tr>
<tr>
<td><strong>Frequency band</strong></td>
<td>880-915/935-960 MHz</td>
<td>824-849/869-894 MHz</td>
<td>824-849/869-894 MHz</td>
<td>810-826/940-956 MHz</td>
</tr>
<tr>
<td><strong>Maximum possible data rate</strong></td>
<td>HSCSD: 57.6 kbit/s</td>
<td>IS-136+: 43.2 kbit/s</td>
<td>IS-95A: 14.4 kbit/s</td>
<td>28.8 kbit/s</td>
</tr>
<tr>
<td><strong>Handover</strong></td>
<td>Hard</td>
<td>Hard</td>
<td>Soft</td>
<td>Hard</td>
</tr>
</tbody>
</table>
(General Packet Radio Service), and IS-95B, are commonly referred as generation 2.5 (2.5G). HSCSD, GPRS and EDGE are all based on the original GSM system. HSCSD is the first enhancement of the GSM air interface: it bundles GSM timeslots to give a theoretical maximum data rate of 57.6 kbit/s (bundling 4x14.4 kbit/s full rate timeslots). HSCSD provides both symmetric and asymmetric services and it is relatively easy to deploy. However, HSCSD is not easy to price competitively since each time slot is effectively a GSM channel.

Following HSCSD, GPRS is the next step of the evolution of the GSM air interface. Other than bundling timeslots, 4 new channel coding schemes are proposed. GPRS provides “always on” packet switched services with bandwidth only being used when needed. Therefore, GPRS enables GSM with Internet access at high spectrum efficiency by sharing time slots between different users. Theoretically, GPRS can support data rate up to 160 kbit/s (current commercial GPRS provides 40 kbit/s). Deploying GPRS is not as simple as HSCSD because the core network needs to be upgraded as well. EDGE uses the GSM radio structure and TDMA framing but with a new modulation scheme, 8QPSK, instead of GMSK, thereby increasing by three times the GSM throughput using the same bandwidth. EDGE in combination with GPRS will deliver single user data rates of up to 384 kbit/s.

1.1.3 Third-Generation Mobile Systems and Beyond

The massive success of 2G technologies is pushing mobile networks to grow extremely fast as ever-growing mobile traffic puts a lot of pressure on network capacity. In addition, the current strong drive towards new applications, such as wireless Internet access and video telephony, has generated a need for a universal standard at higher user bit rates: 3G. Figure 1.2 shows the bit rate requirements for some of the applications that are
predicated for 3G networks (Saussy 2002). Most of the new services require bit rates up to 2 Mbit/s and more.

Because of these drivers, the International Telecommunications Union (ITU) has been developing 3G since 1985. 3G networks are referred as IMT-2000 (International Mobile Telephony) within ITU and UMTS (Universal Mobile Telecommunications Services) in Europe. In ETSI (European Telecommunications Standards Institute), UMTS standardisation started in 1990.

Figure 1.2 Bit rate requirements for some 3G applications

Third generation systems are designed for multimedia communications: person-to-person communication can be enhanced with high quality images and video. Also, access to information and services on public and private networks will be enhanced by the higher data rates and new flexible communication capabilities of these systems. Third generation systems can offer simultaneous multiple services for one user and services with different Quality of Service (QoS) classes.
The main objectives of the IMT-2000 systems can be summarised as:

- Full coverage and mobility for 144 kbit/s, preferably 384 kbit/s.
- Limited coverage and mobility for 2 Mbit/s.
- Provides both symmetric and asymmetric data transmission.
- Provides both circuit switched and packet switched connections.
- Capable of carrying Internet Protocol (IP) traffic.
- Global roaming capabilities.
- High spectrum efficiency compared to existing systems.
- High flexibility to introduce new services.

The bit-rate targets have been specified according to the integrated services digital network (ISDN). The 144 kbit/s data rate provides the ISDN 2B+D channel configuration, 384 kbit/s provides the ISDN H0 channel, and 1.92 Mbit/s provides the ISDN H12 channel. Figure 1.3 shows the relation between bit rates and mobility for the different systems.

![Figure 1.3 User bit rate versus coverage and mobility](Holma and Toskala 2002).
1.1.3.1 Air Interface and Spectrum Allocation

Within the IMT-2000 framework, several different air interfaces are defined for 3G systems, based on either CDMA or TDMA technology. Currently, there are five interfaces agreed by ITU and being under standardisation as shown in Figure 1.4.

![IMT-2000 air interfaces](image1.png)

**Figure 1.4** IMT-2000 air interfaces (Scrase 2002)

Among these interfaces, WCDMA has been adopted as the radio access technology of UMTS; it is also to be used in Asia, including Japan and Korea. Multicarrier CDMA (cdma2000) can be used as an upgrade solution for the existing IS-95.

![Spectrum allocation in different countries](image2.png)

**Figure 1.5** Spectrum allocation in different countries (Scrase 2002)
In 1992, the World Administrative Radio Conference (WARC) allocated spectrum bands 1885-2025 MHz and 2100-2200 MHz for IMT-2000 systems, and in WRC2000, two further spectrum bands 1710-1885 MHz and 2500-2690 MHz were added. However, different countries have their own usage because of the different choices of 3G air interface and the different existing 2G systems. Figure 1.5 shows the spectrum allocations for 3G systems in different countries and Table 1.2 shows the expected frequency bands and geographical areas where different air interfaces are likely to be applied (Holma and Toskala 2002).

Table 1.2 Expected spectrums and air interfaces for providing 3G services

<table>
<thead>
<tr>
<th>Area</th>
<th>Frequency band</th>
<th>Air Interface</th>
</tr>
</thead>
<tbody>
<tr>
<td>Europe</td>
<td>IMT-2000 band</td>
<td>WCDMA</td>
</tr>
<tr>
<td></td>
<td>GSM 1800 band</td>
<td>EDGE</td>
</tr>
<tr>
<td>Americas</td>
<td>In the existing bands that are already used by second generation systems.</td>
<td>EDGE, WCDMA, and multicarrier CDMA (cdma2000)</td>
</tr>
<tr>
<td>Asia</td>
<td>IMT-2000 band</td>
<td>WCDMA</td>
</tr>
<tr>
<td></td>
<td>GSM 1800 band</td>
<td>EDGE</td>
</tr>
<tr>
<td>Japan</td>
<td>IMT-2000 band</td>
<td>WCDMA</td>
</tr>
</tbody>
</table>

1.1.3.2 3G Systems and Beyond

Currently 3G technologies are starting to be launched commercially; for example, Hutchison launched 3G in the UK in 2002. Recently, systems beyond 3G (B3G systems) are attracting more and more
attention. Many international Forums such as ITU-R WP8F Vision Group and the EU initiated Wireless World Research Forum (WWRF) are undertaking research on B3G systems. In 2002, ITU-R WP8F held the 9th meeting, subtitled “System Capabilities for system beyond 3G”. Higher bit rate with higher mobility is expected to be required for future new applications.

Meanwhile, fourth-generation mobile networks branded 4G have already been proposed. Although there is no uniform definition about what 4G is and what exactly marks the generation, the vision for the 4G mobile networks is developing, with different views being taken in different areas around the world. In Asia, 4G is being foreseen as an air interface that could support up to 100 Mbit/s for high mobility and up to 1 Gbit/s for low mobility. In the US, 4G is expected to be the combination of Wireless Local Access Network (WLAN) and IEEE 802.201. In Europe, the understanding of 4G is a network of networks, which includes multiple interworking networks and devices. In this kind of network, the following different technologies might coexist with seamless interworking being supported between them.

- Cellular Mobile (2/2.5/3G)
- Wireless LAN (IEEE802.11x)
- Personal Area Networking (Bluetooth)
- Digital Broadcasting (video, audio, DVB, DAB)
- Home Entertainment Wireless Networking
- Multi-Modal Services

In the future, wireless communication is going to move towards universal communication that uses a very flexible networking infrastructure and dynamically adapts to the changing requirements.
1.1.4 Overview Of CDMA Technology

Digital communications systems are designed to maximise capacity utilisation. From Shannon’s channel capacity principle expressed as (1.1), it is obvious that the channel capacity can be increased by increasing the channel bandwidth.

$$C = B \log_2 \left(1 + \frac{S}{N}\right) \tag{1.1}$$

where $B$ is the bandwidth (Hz), $C$ is the channel capacity (bit/s), $S$ is the signal power and $N$ is the noise power. Thus, for a particular $S/N$ ratio (Signal to Noise Ratio: SNR), the capacity is increased if the bandwidth used to transfer information is increased. CDMA is a technology that spreads the original signal to a wideband signal before transmission. CDMA is often called as Spread-Spectrum Multiple Access (SSMA). The ratio of transmitted bandwidth to information bandwidth is called the processing gain $G_p$ (also called spreading factor).

$$G_p = \frac{B_t}{B_i} = \frac{B}{R} \tag{1.2}$$

where $B_t$ is the transmission bandwidth, $B_i$ is the bandwidth of the information bearing signal, $B$ is the RF bandwidth and $R$ is the information rate. Relating the $S/N$ ratio to the $E_b/I_0$ ratio, where $E_b$ is the energy per bit, and $I_0$ is the noise power spectral density, leading to:

$$\frac{S}{N} = \frac{E_b \cdot R}{I_0 \cdot B} = \frac{E_b}{I_0 \cdot G_p} \tag{1.3}$$
Therefore, for certain $E_b/I_0$ requirement, the higher the processing gain, the lower the S/N ratio required. In the first CDMA system, IS-95, the transmission bandwidth is 1.25 MHz. In WCDMA system, the transmission bandwidth is about 5 MHz. In CDMA, each user is assigned a unique code sequence (spreading code) that is used to spread the information signal to a wideband signal before being transmitted. The receiver knows the code sequence for that user, and can hence decode it and recover the original data.

A mobile communication network is a multi-user system, in which a large number of users share a common physical resource to transmit and receive information. Multiple access capability is one of the fundamental components. The spectral spreading of the transmitted signal gives the feasibility of multiple access to CDMA systems.

Figure 1.6 shows three different multiple access technologies: TDMA, FDMA and CDMA. In FDMA, (Frequency Division Multiple Access), signals for different users are transmitted in different channels each with a different modulating frequency; in TDMA, (Time Division Multiple Access), signals for different users are transmitted in different time slots. With these two technologies, the maximum number of users who can share the physical channels simultaneously is fixed. However, in CDMA, signals for different users are transmitted in the same frequency band at the same time. Each user’s signal acts as interference to other user’s signals and hence the capacity of the CDMA system is related closely to the interference level: there is no fixed maximum number, so the term ‘soft capacity’ is used. Figure 1.7 shows an example of how 3 users can have simultaneous access in a CDMA system.
At the receiver, user 2 de-spreads its information signal back to the narrow band signal, but nobody else’s. This is because that the cross-correlations between the code of the desired user and the codes of other users are small: coherent detection will only put the power of the desired signal and a small part of the signal from other users into the information bandwidth. The processing gain, together with the wideband nature of the process, gives benefits to CDMA systems, such as high spectral efficiency and soft capacity. However, all these benefits require the use of tight power control and soft handover to avoid one user’s signal cloaking the communication of others.
Signal to Interference Ratio (SIR) is similar to the quantity known as Signal to Noise Ratio (SNR) in communications and signal processing applications. It is defined as in equation 1.4 below.

\[
SIR = \frac{\text{Signal Power}}{\text{Total Interference Power}}
\]  

Figure 1.8 shows the two types of interference which can occur in a CDMA system. A frequency division duplex (FDD) link is used to communicate between the mobile terminal and the base station. All uplink connections are on one frequency band and the downlink connections on another. Figure 1.8a highlights interference caused by the uplink channels of mobiles in the vicinity and Figure 1.8b shows downlink interference to a mobile caused by neighbouring base stations. This type of interference is unique to a CDMA system due to the usage of same frequency by all the cells.

Figure 1.8  Interference experienced by a base station and a mobile host  
(Steele 1992)

The SIR equation 1.4 can be expanded as follows:

\[
SIR = SF \cdot \frac{P_i}{I_{int,ra} + I_{int,er} + P_N}
\]  

(1.5)
where,

- $P_r$ is the received signal strength.
- $P_N$ is the thermal noise power assumed equal to -99dBm in downlink, and -103dBm in uplink.
- $I_{\text{inter}}$ is the sum of signal powers received from other cells.
- $I_{\text{intra}}$ is the sum of signal powers due to other transmissions within the same cell.
- $SF$ is the spreading factor for a given call type.

### 1.1.5 Features of WCDMA

Wideband-CDMA (WCDMA) has been adopted by UMTS as the multiple access technology and it is also referred to as UMTS terrestrial radio access (UTRA). This section introduces the principles of the WCDMA air interface. Special attention is drawn to those features by which WCDMA differs from GSM and IS-95. Table 1.3 summarises the main parameters related to the WCDMA air interface. Some of the items that characterise WCDMA are:

- WCDMA is a wideband CDMA system. User information bits are spread over a wide bandwidth (5 MHz) by multiplying with spreading codes before transmission and are recovered by decoding in the receiver (Ojanpera and Prasad 1998).
- The chip rate of 3.84 Mchip/s used leads to a carrier bandwidth of approximately 5 MHz. In GSM, carrier bandwidth is only 200 kHz. Even in narrowband CDMA systems, such as IS-95, the carrier bandwidth is only 1.25
MHz. The inherently wide carrier bandwidth of WCDMA supports high user data rates and also has certain performance benefits, such as increased multipath diversity.

- WCDMA supports highly variable user data rates; in other words the concept of obtaining Bandwidth on Demand (BoD) is well supported. Each user is allocated frames of 10 ms duration, during which the user data rate is kept constant. However, the data capacity among the users can change from frame to frame.

- WCDMA supports two basic modes of operation: Frequency Division Duplex (FDD) and Time Division Duplex (TDD). In FDD mode, separate 5MHz carriers are used for the uplink and downlink respectively, whereas in TDD only one 5 MHz is time-shared between uplink and downlink.

- WCDMA supports the operation of asynchronous base stations. Unlike the synchronous IS-95 system, there is no need for a global time reference, such as a GPS, so making deployment of indoor and micro base stations easier.

- WCDMA employs coherent detection on uplink and downlink based on the use of pilot symbols or common pilot. In IS-95 coherent detection is only used on the downlink. The use of coherent detection on uplink will result in an overall increase of coverage and capacity on the uplink. This makes the downlink more likely to be the bottleneck of the whole system.

- The WCDMA air interface has been crafted in such a way that advanced CDMA receiver concepts, such as multiuser detection (MUD) (Verdu 1986, Ojanpera et al 1998, Juntii and
Latvaaho 2000) and smart adaptive antennas (Lee and Arnott 2001), can be deployed by the network operator as a system option to increase capacity and/or coverage. In most second generation systems no provision has been made for such concepts.

- WCDMA is designed to be deployed in conjunction with GSM. Therefore, handovers between GSM and WCDMA are supported.

**Table 1.3 Main WCDMA parameters**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multiple access method</td>
<td>DS-CDMA</td>
</tr>
<tr>
<td>Duplexing method</td>
<td>FDD/TDD</td>
</tr>
<tr>
<td>Base station synchronisation</td>
<td>Asynchronous operation</td>
</tr>
<tr>
<td>Chip rate</td>
<td>3.84 Mcps</td>
</tr>
<tr>
<td>Frame length</td>
<td>10 ms</td>
</tr>
<tr>
<td>Service multiplexing</td>
<td>Multiple services with different quality of service requirements multiplexed on one connection</td>
</tr>
<tr>
<td>Multirate concept</td>
<td>Variable spreading factor and multicode</td>
</tr>
<tr>
<td>Detection</td>
<td>Coherent using pilot symbols or common pilot</td>
</tr>
<tr>
<td>Multiuser detection, Smart antennas</td>
<td>Supported by the standard, optional in the implementation</td>
</tr>
</tbody>
</table>
1.1.6 Radio resource Management and its Functionalities

Radio Resource Management (RRM) in 3G networks is responsible for improving the utilisation of the air interface resources. The objectives of using RRM can be summarised as follows:

- Guarantee the QoS for different applications.
- Maintain the planned coverage.
- Optimise the system capacity.

In 3G networks, pre-allocating resource and over-dimensioning the network are not feasible any more because of the unpredictable need and the variable requirements of different services. Therefore, radio resource management is composed of two parts: radio resource configuration and re-configuration. Radio resource configuration is responsible for allocating the resource properly to new requests coming into the system so that the network is not overloaded and remains stable. But, as congestion might occur in 3G networks because of the mobility of users, radio resource re-configuration is responsible for re-allocating the resource within the network when load is building up or congestion starts to appear. It is responsible for returning the overloaded system quickly and controllably back to the targeted load.

Radio resource management can be divided into power control, handover, admission control and load control functionalities. Figure 1.9 shows the typical locations of RRM functionalities within a WCDMA network.
Figure 1.9  Typical locations of RRM functionalities within a WCDMA network (Holma and Toskala 2002).

i)  Power control

Power control is a necessary element in all mobile systems because of the battery life problem and safety reasons, but in CDMA systems, power control is essential because of the interference-limited nature of CDMA. In GSM slow (frequency approximately 2 Hz) power control is employed. In IS-95 fast power control with 800 Hz is supported in the uplink, but in the downlink, a relatively slow (approximately 50 Hz) power control loop controls the transmission power. In WCDMA fast power control with 1.5 kHz frequency is supported in both uplink and downlink (Ojenpara and Prasad 1998). Tight and fast power control is one of the most important aspects of WCDMA systems. The overall objectives of power control can be summarised as follows:

- Overcoming the near-far effect in the uplink
- Optimising system capacity by controlling interference
- Maximising the battery life of mobile terminals (not considered further)
There are three types of power control in WCDMA systems: open-loop power control, closed-loop power control, and outer-loop power control.

Open-loop power control is used in the UMTS FDD mode for the mobile initial power setting. The mobile estimates the path loss between the base station and the mobile by measuring the received signal strength using an automatic gain control (AGC) circuit. According to this estimate of path loss, the mobile can decide its uplink transmit power. Open-loop power control is effective in a TDD system because the uplink and downlink are reciprocal, but it is not very effective with FDD system because the uplink and downlink channels operate on different frequency bands and the Rayleigh fading in the uplink and downlink is independent. So open-loop power control can only roughly compensate distance attenuation. That is why it is only used as an initial power setting in FDD systems.

Closed-loop power control, also called fast power control in WCDMA systems, is responsible for controlling the transmitted power of the MS (uplink) or of the base station (downlink) in order to counteract the fading of the radio channel and meet the SIR (signal-to-interference ratio) target set by the outer-loop. For example, in the uplink, the base station compares the received SIR from the MS with the target SIR once every time slot (0.666ms). If the received SIR is greater than the target, the BS transmits a Transmit Power Control (TPC) command “0” to the MS via the downlink dedicated control channel. If the received SIR is lower than the target, the BS transmits a TPC command “1” to the MS. Because the frequency of closed-loop power control is very fast it can compensate fast fading as well as slow fading.

Outer-loop power control is needed to keep the quality of communication at the required level by setting the target for the fast closed-loop power control. It aims at providing the required quality: no worse, no
better. The frequency of outer-loop power control is typically 10-100Hz. The outer-loop power control compares the received quality to the required quality. Usually the quality is defined as a certain target Bit Error Rate (BER) or Frame Error Rate (FER). The relationship between the SIR target and the quality target depends on the mobile speed and the multipath profile. If the received quality is better, it means the current SIR target is high enough for guaranteeing the required QoS. In order to minimise the headroom, the SIR target will be reduced. However, if the received quality is worse than the required quality, the SIR target needs to be increased for guaranteeing the required QoS.

ii) **Handover control**

Handover or handoff is an essential component of mobile cellular communication systems. Mobility causes dynamic variations in link quality and interference levels in cellular systems, sometimes requiring that a particular user change its serving base station. This change is known as handover. In WCDMA system soft handoff is employed. With the soft handoff functionality the handset can communicate simultaneously with two or more cells in two or more base stations. This flexibility in keeping a connection open across base stations results in fewer dropped calls. Soft handoff enables the handset to maintain the continuity and the quality of the connection while moving from one cell to another. During soft handoff the handset momentarily adjusts its power to the base station that requires the smallest amount of transmit power and the preferred cell may change very quickly. In a well designed radio network 30% to 40% users are regularly in soft handoff.
iii) Admission control

If the air interface loading is allowed to increase excessively, the coverage area of the cell is reduced below the planned values (so called “cell breathing”), and the QoS of the existing connections cannot be guaranteed. The reason for the “cell breathing” phenomenon is because of the interference-limited feature of CDMA systems. Therefore, before admitting a new connection, admission control needs to check that admitting the new connection will not sacrifice the planned coverage area or the QoS of existing connections. Admission control accepts or rejects a request to establish a radio access bearer in the radio access network. The admission control functionality is located in Radio Network Controller (RNC) where the load information of several cells can be obtained. The admission control algorithm estimates the load increase that the establishment of the bearer would cause in the radio access network. The load estimation is applied for both uplink and downlink. The requesting bearer can be admitted only if the admission controls in both directions admit it, otherwise it is rejected because of the excessive interference that it adds to the network. Generally, the admission control strategies can be divided into two types: wideband power-based or interference based admission control strategy and throughput-based admission control strategy.

The new user is not admitted if the new resulting total interference level is higher than the threshold value:

\[ I_{\text{total,old}} + \Delta I > I_{\text{th}} : \text{Reject} \]
\[ I_{\text{total,old}} + \Delta I < I_{\text{th}} : \text{Admit} \]

The threshold value is the same as the maximum uplink noise increase and can be set by radio network planning.
In throughput-based admission control strategy the new requesting user is not admitted into the radio access network if the new resulting total load is higher than the threshold value:

$$\rho_{\text{total, old}} + \Delta \rho > \rho_{\text{th}} : \text{Reject}$$
$$\rho_{\text{total, old}} + \Delta \rho < \rho_{\text{th}} : \text{Admit}$$

It should be noted that as the admission control is applied separately for uplink and downlink, different admission control strategies can be used in each direction.

iv) Load Control (Congestion Control)

One important task of the radio resource management functionality is to ensure that the system is not overloaded and remains stable. If the system is properly planned, and the admission control works well, overload situations should be exceptional. However, in mobile networks, overload somewhere is inevitable because the radio resource cannot be pre-allocated within the network. When overload is encountered, the load control, also called congestion control, returns the system quickly and controllably back to targeted load, which is defined by the radio network planning. The possible load control actions in order to reduce or balance load are listed below:

- Deny downlink power-up commands received from the MS.
- Reduce the uplink $E_b/I_0$ target used by uplink fast power control.
- Change the size of soft handover zone to accommodate more users.
- Handover to another WCDMA carrier (inter-frequency handover).
- Handover to overlapping network (another UMTS network or GSM network).
- Decrease bit rates of real-time users, e.g. AMR speech codec.
- Reduce the throughput of packet data traffic (non real-time service).
- Drop calls in a controlled way.

The first two in the list are fast actions that are carried out within the BS. These actions can take place within one timeslot, i.e. with 1.5 kHz frequency, and provide prioritisation of the different services. The third method, changing the size of soft handover zone is especially useful to a downlink-limited network.

The other load control actions are typically slower. Inter-frequency handover and inter-system handover can overcome overload by balancing the load. The final action is to drop real-time users (i.e. speech or circuit switched data users) in order to reduce the load. This action is taken only if the load of the whole network remains very high even after other load control actions have been effected in order to reduce the overload. The WCDMA air interface and the expected increase of non real-time traffic in third generation networks give a large selection of possible actions to handle overload situations, and therefore the need to drop real-time users to reduce overload should be very rare.
1.2 SURVEY OF CALL ADMISSION CONTROL ALGORITHMS FOR WCDMA NETWORKS

A survey of existing call admission algorithms and some remarks about the strengths and weaknesses of these proposals, based on experimental research reported in the literature, are presented in this section. This provides a basis for the development and evaluation of the modifications proposed in this thesis.

1.2.1 Different Call Admission Schemes

For the purposes of review the admission schemes in existence can be classified as either Interactive or Non-Interactive scheme.

1.2.1.1 Interactive Call Admission Schemes

Ideally a call admission scheme accepts a new call only if the closed-loop Power Control mechanism is able to reach a new equilibrium where all connections observe a target signal to interference ratio to ensure good quality. Interactive Call Admission scheme behavior is very close to ideal admission control because it allows new connections to transmit for a trial period during which it takes measurements to determine whether the connection can be tolerated. Unfortunately the procedure required for such a scheme is too complex considering that during the trial period the scheme must ensure the new call does not affect the quality of the ongoing calls. Also taking measurements and making decisions with Interactive Admission schemes can be very time consuming. The other drawback is its inability to work with inactive connections. Interactive schemes can only work with always active connections and cannot exploit discontinuous transmission, which is very important in UMTS. Andersin et al (1997), Kim (2000) are
examples of Interactive Call Admission Schemes and provide further detail on the subject.

1.2.1.2 Non-Interactive Call Admission Schemes

Unlike Interactive schemes, Non-Interactive schemes only estimate the network load by measuring a few system parameters. The decisions on call admission are based on the estimates. The total interference measured at the base station is generally considered as a good load index, since the ability of the power control mechanism to keep SIR at the target level depends on the interference level. The measured interference includes both intra-cellular and inter-cellular interference. Therefore the admission decision can be based on interference experienced in the cell of the base station as well as in neighboring cells. The measured values are compared with a threshold and is only accepted if the threshold is not exceeded.

The acceptance thresholds are tuned to limit the dropping probability. The simple Receive Power based Admission Control schemes do not consider the additional load due to the new call. The threshold tuning must take into account that the load increase is highly varying depending on mobile terminal position and propagation conditions towards its Base Station and others. The acceptance threshold must be kept low in order to tolerate the worst possible scenarios and to minimize dropping probability. As a result the acceptance probability will be much lower in Non-interactive schemes than those of near ideal schemes. Examples are given in Huang and Yates (1996), Knutsson et al (1997 and Knutsson et al (1998).
1.2.2 Call Admission Control in CDMA System

Signal to Interference ratio (SIR) based call admission control algorithm is proposed in Liu and Zarki (1994). Residual capacity is introduced as the additional number of initial calls a base station can accept such that system-wide outage probability will be guaranteed to remain below a certain level. Since the acceptance of handover calls in a DS-CDMA cellular system will result in reducing the system wide interference, from the interference point of view, it can be viewed that handover calls to a cell will not request residual capacity from the cell. The residual capacity at each cell is updated dynamically according to the reverse link SIR measurements at the base station.

Two distributed SIR-based CAC algorithms for the DS-CDMA cellular system are proposed and analysed in their work. The first algorithm is a totally localized algorithm, and the CAC decision is solely based on the SIR measurement at the local base station. The second algorithm utilizes the local SIR measurement and also the SIR measurement of the adjacent cells (immediate neighbour cells). In both the algorithms the voice activity factor of the voice call and the mobility of the users are not modelled. The performance of this algorithm was compared with that of a fixed call admission control scheme under both homogeneous and hot spot traffic loadings.

The received power based call admission control(RPCAC) technique for UMTS uplink is proposed in Simone Radana and Antonio Capone (2002). The performance of their predictive algorithm with the simple receive power based call admission control algorithm for both single cell and multi cell was compared in their work.
Call admission policies for UMTS is given in Nikos Dimitriou et al (2000). The problem of Call Admission control for WCDMA UMTS is analysed and the different factors that affect the system soft capacity are presented. A way of accessing the sector load and determining the system residual capacity are proposed and the mathematical analysis for the QoS requirement of all the users in a sector are presented. The Call Admission control presented in their paper is based on the transmitted power by the user terminal which attempt to mitigate the propagation channel impairments and the effect of interference generated by the users in the same and other sectors. Two CAC policies are studied and analysed in their work. In the first one, the threshold was fixed (fixed case) and in the second one the threshold was varied according to the instantaneous dropping rate per service. The performance of fixed and dynamic CACs are compared with No CAC.

The safety margin of admission thresholds used by Non-Interactive schemes can be reduced by using a scheme to estimate the additional interference due to the new call. Predictive schemes discriminate between calls requested by mobile terminals with different propagation conditions. This approach produces a non-uniform accepted traffic distribution where terminals close to the base station are more likely to be accepted. This is similar to ideal call admission, which exploits this effect to increase accepted traffic. Interference increase estimation algorithms are given in Holma and Laakso (1999) and Outes et al (2001).

Call Admission in Power controlled CDMA Systems are studied in Huang and Yates (1996). They analysed two different CAC schemes in their work. In the transmitter power Call admission control (TPCAC), a new call is blocked when this call causes ongoing calls to transmit at maximum power. It admits a call, if all ongoing calls can maintain acceptable SIR. In the received power CAC, a new call is blocked at the base station, when the total power
measured at the base station exceeds the threshold. In TPCAC, since the new calls participate in the iterative power control before a decision is made, this is a form of interactive admission control. There is a trade off in the choice of number of iterations. With large number of iterations, the system is more likely to take a correct decision, but the set up phase will last longer which will increase the implementation complexity. In RPCAC, the new user will be accepted at the base station as long as the total received power at the base station is less than a threshold.

The call admission scheme with total receive power as thresholds for multi rate traffic in CDMA systems is proposed in Kuenyoung Kim and Youngnam Han (2000). Multiple thresholds are used in the admission control scheme based on the priority of the traffic.

The impact of mobility on CAC in WCDMA systems is analysed in mobility aware interference based CAC proposed by Badia et al (2002). They considered four different mobility classes like stationary, pedestrian, slow vehicle and fast vehicle and used different admission thresholds for the different mobility classes. The continuous time under threshold policy, which is one of the dropping policies used to avoid the congestion is studied. If the SIR of the user remains below a threshold $SIR_{th}$ for a specified amount of time, congestion is detected and the user is dropped. This dropping process is unfair and highly dependent on user’s parameter like mobility and call duration time. In this algorithm, the system tracks the speed of the mobile users and it uses a variable admission threshold for received power depending on the users speed. The method used for estimating the speed of the users is not discussed in their work.

In distributed admission control for power controlled cellular wireless systems in Mingbo Xiao et al (2001), the admission control problem
for cellular systems under optimal distributed power control in general without considering any multiple access technique was studied. The mobility of the users and its impact on the performance of the admission control algorithm, performance with multiple traffic classes are not discussed.

1.2.3 Prioritization Schemes to Improve QoS

1.2.3.1 Handoff Prioritization

In the call admission control for CDMA mobile communication systems supporting multimedia services proposed in Wha Sook Jeon and Dong Geun Jeong (2002), two traffic classes with higher priority for handoff calls in each class were considered. The traffic asymmetry between the uplink and downlink was also taken into account. The priority mechanism between call classes is implemented by adequately assigning different SIR thresholds to different call classes. The parameter based on which the difference between SIR thresholds maintained for the different call classes is not discussed.

Provisioning QoS over WCDMA cannot be fulfilled with just proper admission control and efficient scheduling. This is due to mobility of the users, fading channels (Rappaport 2002), high error rates (Ericsson Radio System 2001), low and varying bandwidth, but mainly due to the unexpected handoff requests (Rachidi et al 2004).

Most of the issues mentioned above have been catered for using closed-loop power control mechanisms that operate solely on the basis of channel gain, but that are not aware of QoS requirements of underlying connections. This blind mode of operation does not necessarily yield optimal power utilization, especially when other non-premium connections in the system are willing to be degraded, that is they are capable of adaptation and
willing to have their required bit-rate/power reduced. The issue of unexpected handoffs has been solved using reservation and prediction schemes (Soh and Kim 2003).

The QoS aware Power Control and Handoff Prioritization scheme is proposed in Rachidi et al (2004). This scheme shows that user willingness to be degraded can be used to augment both traditional closed loop control mechanisms for congestion handling, as well as, to improve handoff by reducing the rate of blocking handoff requests.

Subscriber Degrade Descriptor (SDD) is presented in Lataoui et al (2000) for modelling user willingness to be degraded. SDD is a number between 0 - 5 and larger the SDD is the more willing, a user is to be degraded and to eventually be dropped. Rachidi et al (2004) uses SDD together with service classes and the bit-rates as enabling QoS parameters in systems capable of minimizing handoff call blocking and new call blocking.

### 1.2.3.2 Non Real-Time Overload Admission Strategy

The non real-time (NRT) overload strategy is used for admission decisions in Rachidi et al (2004). In the traditional strict admission strategy a new call is accepted in the system at an instant only if the power required by all users do not exceed the total power available and if the QoS requirements of the other users are not lowered. But in the NRT overload admission strategy the base station is allowed to accept connections even if the total power required by all users exceed the available power. Here the NRT connections are backed off and delayed by a scheduler. Specifically, a new connection is accepted in the system at instant ‘t’ if and only if:

$$\sum P_{i\text{RT}}(t) \leq P_{\text{max}}$$  \hspace{1cm} (1.6)
where $P_{i/RT}(t)$ is the power required by existing real-time connection ‘i’, (that is class 1 and 2 connections in the system including eventually the new connection).

$$\sum P_i(t) \leq (1 + \alpha)P_{\text{max}}$$  \hspace{1cm} (1.7)

where $0 < \alpha < 1$ indicates the maximum overload allowed for NRT connections, and ‘i’ spans across all existing connections including the one under the admission decision.

1.2.3.3 Service Specific CAC in WCDMA System

The service specific call admission control in WCDMA System is proposed in Lee and Jun Jo (2004). It contains CDMA capacity evaluation and resource reservation for hand-off calls. This scheme considers the priority of each service class. To simplify the problem, they defined three service classes with priorities given in the following order: real time high data rate service, real time low data rate service and real time voice service. Each service call is classified into either the handoff or new call. Therefore, there are six priority levels in total and the handoff call is given higher priority than the new call.

In WCDMA, it is important to satisfy the required QoS of the calls, which is expressed in terms of SIR. When a new call enters the system, total noise of the system increases and the SIR of all other calls decreases. The system has to satisfy the required SIR of all the calls, including the newly entered calls. Moreover, the system should have codes available for new calls. In WCDMA system, orthogonal variable spreading factor (OVSF) code is employed to serve various data rate services. Even though OVSF code
assignment may encounter the code blocking problem, it can be eliminated by the process of reassigning the codes. Compared to other CAC mechanisms, this scheme considers the OVSF code assignment to the newly admitted calls and determines whether code reassignment is necessary. The performance of this scheme is evaluated in terms of the new call blocking probability and handoff call dropping probability.

1.2.3.4 CAC for Prioritized adaptive multimedia services

A Call admission control for Prioritized adaptive multimedia services in wireless/Mobile networks is proposed in Sooyeon Kim et al (2000). In this work, the bandwidth of a call takes a value from a set of discrete values depending on situations. The prioritization among multiple classes of services is taken into account; that is, bandwidths of lower priority calls are preferably reduced in overload condition. A threshold type call admission control algorithm is used and a nonlinear programming (NLP) model is formulated to find out threshold values for optimal solution.

A 2-level call admission control scheme using priority to guarantee the consistent QoS for mobile multimedia applications is proposed in Myung Il Kim and Sung Jo Kim (2003). This two level CAC consists of the basic admission and advanced call admission. The basic admission determines the call admission based on bandwidth available in each cell and the advanced admission control determines the admission by utilizing delay tolerance time and priority algorithm. At the basic CAC stage, each call (new call and handoff call) requests required certain bandwidth from the base station. If the available bandwidth of the base station satisfies each call’s request, it assigns the bandwidth to the mobile, otherwise, it blocks the call and passes its control to the advanced call admission control. In Advanced CAC, each cell
requires priority queue to store information such as ID and the priority about the mobile terminal. The call priority is calculated using the equation:

$$\text{Call Priority} = w_1 \cdot DP + w_2 \cdot TP + w_3 \cdot VP$$  \hspace{1cm} (1.8)$$

where DP, TP and VP are the distance priority, traffic priority and velocity priority respectively.

‘w’ is the weight factor.

The performance of this algorithm measured in terms of call blocking probability, call dropping probability and bandwidth utilization, was compared with the performance of Complete sharing policy (CSP), Guard channel policy (GCP) and adaptive guard channel policy (AGCP). In CSP, the new and handoff calls are not distinguished and are admitted as long as available bandwidth of a cell can provide the required bandwidth to the mobile terminal.

In GCP, a fixed amount of bandwidth is reserved for multimedia handoff calls. This scheme reserves as much bandwidth as required by a mobile terminal requesting multimedia services and having high probability of entering a certain cell. This scheme has an advantage of having the low call dropping probability ($P_{hd}$) for multimedia handoff calls by assigning reserved bandwidth to them. However, since the available bandwidth is reserved for multimedia calls, new and non-multimedia handoff calls have high call blocking probability ($P_{nb}$) and call dropping probability ($P_{hd}$). Moreover the reserved bandwidth may turn out to be not used, thus resulting in low bandwidth utilization. In AGCP, the bandwidth reservation for multimedia calls is adaptive based on the network condition. The factors based on which the bandwidth reservation is made was not discussed.
1.2.4 CAC for cellular CDMA systems based on best achievable performance

In Nuaymi (2000), a new CAC algorithm based on best achievable performance was proposed. At each new mobile arrival, the best achievable SIR is computed taking into account the presence of new mobile. The admission decision is based on the result. A margin is introduced in order to prevent link gain variations due to the mobility of the mobile users and multi path fading. The performance of this algorithm was compared with received power based CAC and the role of margin was studied. They found that the high value of margin increases the unnecessary blocking at low loads while a small value lets the system work close to its maximal capacity and hence the mobiles would be in outage for unacceptable percentage of time. They suggested that this algorithm could be fine tuned in order to have the system work close to its maximal capacity with acceptable performance.

1.2.5 CAC for reducing the dropped calls in CDMA Cellular systems

In the CAC algorithm proposed in Yue Ma et al (2000), the admission decision is based on the effective traffic loads for both target cell and the neighbouring cells. They assumed that the target cell in a wireless network is partitioned into two zones: Core zone (CZ) and the soft handoff zone (SHZ). With respect to the base station in the target cell the area immediately adjacent to the SHZ is called the neighbourhood zone (NZ). When the mobile station is in soft handoff phase it is assumed that, at most two base stations are in the diversity reception. When a new call arrives, the base station at the target cell first checks whether it is in the CZ or SHZ. This can be achieved through two ways. One way is to obtain the information by the signal strength. The other is to check the number of pilots in the active set
of the new call. The effective load for the target cell is calculated from the following equation.

\[
\text{Effective Load} = K_c + w_s K_s + w_n K_n
\]  \hspace{1cm} (1.9)

where \(K_c\) (\(K_s\)) – number of calls in the CZ(SHZ) before the possible admission of a new call.

\(K_n\) – number of calls in the neighbouring zone.

\(w_s, w_n\) – weights

If the target cell is already saturated, the new call is blocked. Otherwise if the new call is in the SHZ, the effective load for the other cell that covers SHZ is calculated. If this is less than the threshold, the algorithm continues with the final check for the neighbourhood cells. The call is accepted if the maximum effective load for the adjacent cells is below the threshold.

When the soft handoff call is required, the effective load for the cells that cover the SHZ are calculated. The largest value from these is compared with the predetermined threshold. If it is less, the algorithm continues with calculating the maximum effective load for the adjacent cells. If this is also less than the threshold, the soft handoff call is accepted and diversity combination is applied for receiving the call. If the loads are above the threshold, the soft handoff call requirement is put in a queue if the queue is not full, otherwise this call is denied. If the waiting time in the queue exceeds the timeout limit, the call is dropped. They introduced the idea of soft guard channel to prioritize the handoff calls in CDMA system since there are no physical resources. A certain amount of traffic capacity is exclusively reserved for the handoff calls. They have not discussed much about whether
they take the reserved capacity as a constant or variable and also they have not analysed the impact of this on the performance of the admission control algorithm.

1.2.6 Resource Estimation and Reservation Schemes to Improve QoS

The performance of a resource estimation and call admission algorithm for Wireless Multimedia Networks using the shadow cluster concept is analyzed in Levine et al (1997). An accurate determination of the amount of resources that a base station must reserve (to maintain a certain call dropping probability) is an important issue in future wireless networks. In contrast to current systems, future wireless networks will support a wide range of applications with diverse bandwidth requirements. Also in future systems, the demand on wireless bandwidth within a cell may change abruptly in a short period of time (Nanda and Goodman 1997), as for example, when several video or high data rate users enter or leave a cell at the same time. In contrast, in current systems the bandwidth demand usually varies gradually, and hence it is much easier to handle. Moreover, future wireless networks may provide customized QoS parameters on a per call and/or on a service basis, enabling users to select a level of service according to a pricing plan.

In order to maintain an acceptable call dropping probability rate, several schemes (Tekinay and Jabbari 1991) have been proposed to dynamically organize the allocation of bandwidth resources. These schemes consider only limited information from neighbouring cells, and do not specifically consider admission control policies as means to prevent congestion. Issues and relationships between resource reservation, channel assignment, call admission, and traffic intensity have been studied previously (Tajima & Imamura 1998). Admission control policies which determine the
number of new voice or data users for acceptance in a packet radio network are given in Yang and Geraniotis (1994). For these policies, voice users are accepted only if a long term blocking probability is not exceeded, while data users are accepted only if the mean packet delay and the packet loss probability are maintained below certain levels. In Tajima and Imamura (1998), a “flexible” channel assignment scheme is proposed based on the analysis of offered traffic distributions or blocking probabilities. A distributed call admission control procedure is proposed in Naghshineh and Schwartz (1995), which takes into consideration the number of calls in adjacent cells as well as in the cell where a new call request is being made, in addition to the knowledge of the mean call arrival, call departure, and call handoff rates.

None of these schemes consider the individual trends of the users in the wireless network, e.g., position, speed, direction, and bandwidth demands. In this work, they have described a concept called “shadow cluster concept”, a predictive resource estimation scheme which provides high wireless network utilization by dynamically reserving the resources that are needed to maintain the call dropping probability requested by the wireless connection. The shadow cluster concept is a solution to the problems of resource reservation and call admission in a wireless network. It is the first scheme that utilizes real-time information about the dynamics, traffic patterns, and bandwidth utilization of the individual mobile terminals in a network. The shadow cluster scheme is dynamic and proactive, i.e., the amount of resources to be reserved is determined “on-the-fly,” and the control functions on call admissions are aimed at preventing congestion conditions. By using shadow clusters, the number of dropped calls during handoffs can be reduced, and the establishment of new calls that are highly likely to result later in dropped calls can be avoided. Shadow clusters are best suited for wireless networks with small cell sizes (nano, micro, mini), that result in a high number of cell handoffs during the lifetime of the average wireless connection. The shadow
cluster concept is completely distributed. It requires processing overheads in base stations as well as some communication between a mobile terminal’s base station and its neighbours through the wire line network. Since future base stations are likely to be linked by switches which use high-bandwidth optical fiber (Acampora and Naghshineh 1994), and since the shadow cluster algorithms’ cycle time is likely to be in the order of several seconds, the amount of extra information generated by the mechanism should be easily manageable by the wire line network.

In the paper Reservation Strategies for Multi-Media Traffic in a Wireless Environment by Epstein and Schwartz (1995), they have considered a single cell in a wireless network assumed to have an unchanging bandwidth which provides services to heterogeneous users who request service either as new or handoff users. Each user type has both bandwidth and QoS requirements. They took only the QoS issue of call acceptance and dropping in order to minimize the dimensionality of the problem. The QoS requirements dictate that different users be accorded different priorities. It has been considered that several different access policies which span the range between complete sharing (CS) and complete partitioning (CP) and focused on hybrid policies incorporating aspects of both techniques. The CS policy allows all users equal access to the bandwidth available at all times and this results in maximum usage of the available bandwidth. At the same time, it does not differentiate between users of different priority which is problematic from a QoS perspective (Kraimeche Schwartz 1986). Additionally, the narrower bandwidth users achieve access to the system with higher probability than wider bandwidth users which is also “unfair” to the wider bandwidth users. The CP policy, on the other hand, divides up the available bandwidth into separate sub-pools according to user type. This policy allows for more control of the relative blocking/dropping probabilities at the expense of overall usage of the network.
The hybrid policies used in both the wired and wireless environments provide a compromise between the different policies by subdividing the available bandwidth into sections. Part of the bandwidth is completely shared and the other part is completely partitioned. This allows more flexibility in catering to the QoS requirements of the different user types while maintaining higher network usage. They proposed a cost measure which allows easy comparison of different policies at the same time in this work. They used this method to analyze a system consisting of narrowband voice users and wideband images for two different traffic patterns over a range of different loads.

In the mobility based reservation technique proposed by Fei Yu and Leung (2001), the mobility prediction scheme used was based on an optimal and efficient data compression algorithm called LZW algorithm. In their work, the performance improvement of the Call admission control algorithm in the third generation WCDMA system was not analysed.

1.2.7 Adaptive Channel Reservation

An adaptive channel reservation scheme for soft handoff attempts is proposed in a DS-CDMA system (Chang and Sung 2001). This scheme reserves channel capacity only if candidate soft handoff calls require reservations by a threshold mechanism based on pilot signal strength. Thus, the size of reserved capacity is efficiently managed according to the occurrence rate of soft handoff attempts. The performance of the proposed scheme is evaluated by simulation in terms of the blocking probability of new calls and the forced termination probability of soft handoff calls. This scheme can reduce the blocking probability without an increase in the forced termination probability compared to the fixed channel reservation scheme and
the reduction becomes larger as the offered load decreases or the maximum reserved capacity increases. This scheme requires an additional overhead for managing the size of reservation capacity and for measuring the rates of new call blocking and soft handoff failures. The performance of the proposed scheme is analytically evaluated in terms of new call blocking probability and soft handoff failure probability. The performance evaluation does not include all factors of real environments such as the variation of cell size and mobile speed.

In Adaptive Bandwidth Reservation Scheme for High-Speed Multimedia Wireless Networks by Oliveira et al. (1998) a new admission control scheme is proposed to provide high degrees of QoS guarantees for multimedia traffic carried in microcellular, high-speed wireless networks. The proposed scheme combines admission control and bandwidth reservation to guarantee QoS requirements. The proposed scheme considers both local information (e.g., the amount of unused bandwidth in the cell where the user currently resides) and remote information (e.g., the amount of unused bandwidth in the neighbouring cells) to determine whether to accept or reject a connection. Since a mobile user is free to move anywhere, an admission control scheme that relies solely on local information cannot guarantee QoS requirements of a connection throughout its lifetime. The proposed scheme thus uses both local and remote information, and allocates bandwidth in the cell where a connection request originates and reserves bandwidth in all neighbouring cells. When a user moves to a new cell necessitating a call handoff, the reserved bandwidth in the cell that the user is moving into is used to support the handoff connection. In addition, every time a user moves to a new cell, bandwidth is reserved in the new neighbouring cells, and the reserved bandwidth in the cells which are no longer neighbouring to the new cell is released.
Further, this scheme distinguishes real-time traffic and non real-time traffic, and reduces the bandwidth assigned to non real-time connections to provide higher quality of service to real-time connections if necessary. This scheme also adjusts the amount of reserved bandwidth based on the current network conditions. This is done by measuring the average connection-dropping probability (i.e., the probability that handoff connections are dropped due to lack of bandwidth) and the reserved bandwidth usage (i.e., how much of the reserved bandwidth is actually being used) and adjusting the amount of reserved bandwidth accordingly. Therefore, this scheme adapts to various network load conditions.

The performance of adaptive resource allocation for multimedia quality of service (QoS) support in broadband wireless networks is analysed in Lei Huang et al (2004). A service model consisting of three service classes with different handoff-dropping requirements was presented. Appropriate call-admission control and resource-reservation schemes were developed to allocate resources adaptively to the real-time service classes with a stringent delay bound. Based on a multidimensional model analysis, simulations were conducted to evaluate the system performance. Through the simulation they have shown that their proposed system could satisfy the desired QoS of multimedia applications under different traffic loads, while achieving high utilization.

The QoS adaptation algorithm was utilized in both their call admission mechanism and the Power Control mechanism of Rachidi et al (2004). This algorithm resolves congestion in two phases. The two phases are applied differently in case of congestion handling and in case of handoff admission. The QoS profile carried by each user comprises of the traffic class, SDD, and the bit-rate.
• **The Degradation Phase**

This phase is based on the SDD. Iteratively, the active user with the highest SDD is the user that degraded in terms of its bandwidth requirements. Calls are degraded according to Table 1.4.

• **Dropping Phase**

The dropping phase takes place when willing connections are degraded, but congestion persists. Dropping is based on:

\[ F_i(t) = SDD_i P_i(t) \]  

(1.10)

where \( P_i(t) \) is the power required by connection \( i \) at time \( t \). Calls with the highest SDD are the first to be dropped. \( F_i(t) \) will be large for calls with high bandwidth requirements and high SDD values.

<table>
<thead>
<tr>
<th>Original Bit-rate</th>
<th>Degraded Bit-rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>384 Kbps</td>
<td>144 Kbps</td>
</tr>
<tr>
<td>144 Kbps</td>
<td>64 Kbps</td>
</tr>
<tr>
<td>64 Kbps</td>
<td>16 Kbps</td>
</tr>
<tr>
<td>16 Kbps</td>
<td>Not Degraded Further</td>
</tr>
</tbody>
</table>

**Table 1.4 Degradation Schemes**

The QoS adaptation algorithm is invoked to provide the necessary bandwidth for handoff requests that normally would not be accepted due to the lack of resources. Two cases are distinguished depending on the class of
service. RT connections are accepted into the network during congestion by first degradation and then by dropping lower priority users. The NRT connections will not be accepted by dropping other users.

1.3 SUMMARY

An overview of the mobile communication networks and some of the issues related to call admission control and resource management schemes, important issues such as power control, soft handoff, SIR and causes of congestion are treated in detail. The main approaches mentioned in the literature on Call Admission Control are presented. The main advantages and drawbacks of the various CAC schemes are also highlighted.

In Chapter 2 the proposed mobility based, parameters measurement based and fuzzy model based dynamic CAC schemes in WCDMA system are described. Two different mobility models are considered and the resource reservation is done based on mobility prediction by using these models in the mobility based CAC Schemes. In the parameters measurement based dynamic call admission control, a service model which consists of three service classes with different handoff-dropping requirements is considered. The resource reservation is made adaptive in order to meet the target handoff dropping probability for the real time services with stringent delay bound. The non real time traffic serviced by the best effort model is considered. In the Fuzzy model based Call Admission Control for multi-class traffic, the mobility information of the new user requesting connection and already existing users, the type of service request (real time service or non-real time service) and the load factor are taken into account. The load factor is calculated from both intra-cell interference and the inter-cell interference at the base station. In the wireless systems, the decision made by the call admission controller is to be based on the imprecision and uncertain measurements due to user mobility,
dynamic QoS requirements and varying channel conditions. This problem can be overcome by application of fuzzy logic in the admission control scheme which provides a good solution to the development of a call admission control scheme.