國立交通大學

電子工程學系 電子研究所碩士班

碩士論文

第三代UMTS系統於異質系統交接中

基於系統容量的壓縮模式控制方法

Capacity-based Compressed Mode Control Algorithm for Inter-System Handover in UMTS

研 究 生：張正達

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摘要

第三代手機的開發正值巔峰，而佈建之後於其他異質系統(GSM,WLAN)的交接轉換將成為一個重要的議題。由於不同系統的射頻介面皆不一致，通用行動通訊系統提出一種方法稱為壓縮模式，將中斷連線製造一段期間來量測其他系統的強度。執行壓縮模式需要提升傳輸功率來加強加速傳輸資料的可靠性。然而此提升的功率將會導致系統的效能嚴重影響，此篇論文則提出一個基於負荷量的壓縮模式，在基於能成功量測異質系統的前提下，將壓縮模式對負荷量所造成的影響降到最低。此篇內容亦建立一個通用行動通訊系統的模擬平台來模擬實際對異質系統交接的情況。而此模擬平台也成功地驗證提出的新壓縮模式演算法能確實的減少輸出功率的消耗並改善負荷量的降低。而且在量測異質系統載子的品質也都能達到令人滿意的成果。所以若實際採用此提出的演算法將可以大大提升壓縮模式的效能及效率。

Capacity-based Compressed Mode Control Algorithm for Inter-System Handover in UMTS

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Abstract

The multimedia services and the high data rate transmissions have become more popular. Therefore, the third generation is developed to overlap with the existing second generation systems. To have a smooth migration in between two systems, the Inter-system handover is one of the key features in the third generation systems. The compressed mode, with variable transmission gaps and power levels, is standardized to support the inter-frequency/system handover. To minimize the use of system resources while maintaining the border-cell handover quality, a capacity-based compressed mode algorithm is proposed. Considering the tradeoff between the capacity and the connection quality, the algorithm can adaptively manages the compressed mode operation based on the potential impacts on the capacity and the efficiency of the compressed mode measurement. A simulation platform is established based on 19 UMTS cells surrounded by GSM Sea to evaluate the performance. The simulation results shows that appropriate scheduled compressed mode can reduce the power consumption and enhance the capacity while maintaining the quality of the measurements of other systems.

誌謝

轉眼間兩年的研究生生涯呼之而過，回想兩年前毅然決然的從控制的領域轉出，去探尋有趣的新領域，很幸運地能遇到黃經堯教授，帶領我進入這熱門的無線網路聖殿，一路上從基礎重新打紮，並一步一步往高深的學問鑽研，如今我願意將我小小的成果分享給所有在這學問大道上的先賢後進。

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Chapter 1
Introduction

1.1 Overview of the Modern Cellular Systems

In Taiwan, the most popular second generation cellular system is Global System for Mobile Communications (GSM) system [1]. GSM is also widely used even in the world, especially in Europe and Asia. The principal technology in the specification is the air interface based on the hybrid frequency-division / time-division multiple access. Each GSM carrier has 200 kHz bandwidth and occupies one of eight time slots. The maximum transmission rate is 9.6 kbps and the services can only support speech and short message service (SMS). With the demands of multimedia services, the advanced systems are designed to improve the GSM system. Figure 1-1 shows the evolution of cellular systems [2]. High Speed Circuit Switch Data (HSCSD), General Packet Radio Service (GPRS), and Enhanced Data-rate for GSM Evolution (EDGE) are sequentially proposed based on the GSM air interface and infrastructure. Universal Mobile Telecommunications System (UMTS) changes the air interface to code division multiple access (CDMA) and then enters the third generation age. Moreover, High Speed Downlink Packet Access (HSDPA) is suggested for Beyond 3G (B3G) to enhance the transmission rate over the UMTS system.

HSCSD improves the speech coder to support new speed of 14.4 kbps. By using multiple time slots transmission, the maximum transmission rate can extend to 57.6 kbps. GPRS transmit the packet switch data to replace the old circuit switch date. The packet switch connection doesn't require any dedicated end-to-end connection. It only uses network resources and bandwidth when data is actually being transmitted. This means that a given amount of radio bandwidth can be shared efficiently and simultaneously among many users. GPRS transmit a packet data service using TCP/IP and X.25 to offer speeds up to 45-160 kbps. EDGE reuses the GSM carrier bandwidth and time slot structure and changes the modulation scheme. The transmission rate substantially increases to 384 kbps at most. The advantages of EDGE include fast availability, reuse of existing GSM infrastructure, and enabling existing 2G system to deliver 3G service in existing spectrum bands.

UMTS implements Wideband CDMA (WCDMA) over 5 MHz and increases the transmission rate up to 384 kbps. With combining many physical data channel, the maximum data rate can support up to 2 Mbps. HSDPA is a packet-based data service and use enhanced technologies including Adaptive Modulation and Coding (AMC), Multiple-Input Multiple-Output (MIMO), Hybrid Automatic Request (HARQ), fast cell search, and advanced receiver design. Thus, the transmission rate can up to 10 Mbps over the same bandwidth in UMTS.

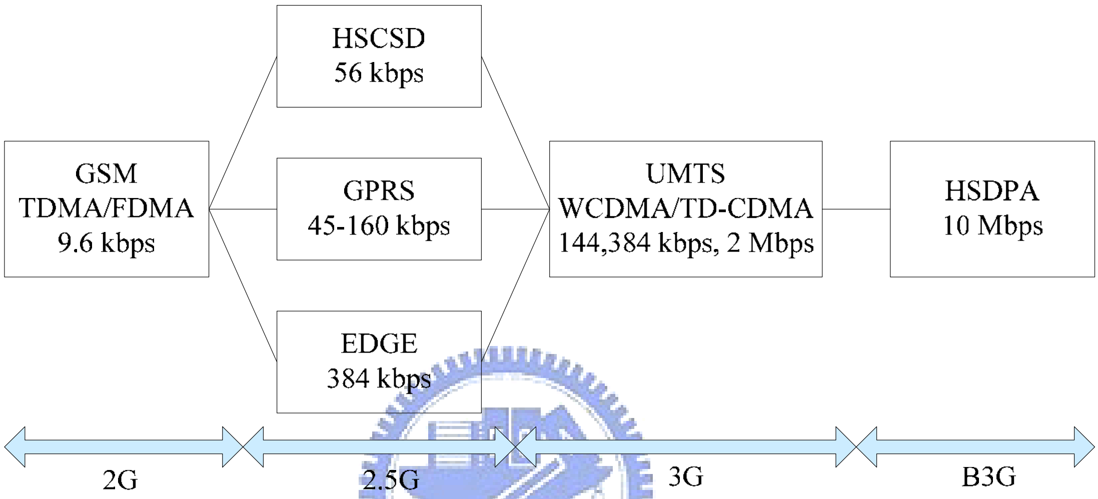


Figure1- 1. The evolution of cellular systems

The UMTS system [3] architecture is depicted in Figure1-2. The UMTS Terrestrial Radio Access Network (UTRAN) handles all radio-related functionality. It consists of two elements, Node B and Radio Network Controller (RNC). The RNC owns and controls the radio resources. The Core Network (CN) is responsible for switching and routing calls and data connections to external networks. Home Location Register (HLR) is the database located in the user's home system that store the user's service profile. Mobile Switching Center (MSC) and Visitor Location Register (VLR) are the switch and database that serves the mobile in its current location for circuit switched services. Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) are the switch and the gateway used for packet switched services. MSC is upgraded from the existing GSM network backbone and connects to circuit switched networks, Public Switched Telephone Network (PSTN) or Integrated Service Digital Network (ISDN). SGSN and GGSN are upgraded from GPRS network backbone and connect to packet data networks. UMTS only separates the architecture by UTRAN for the different air interface. Other elements are based on an evolved GSM core network, provided backward compatibility the GSM in terms of network protocol and interface. The core network can support both GSM and UMTS services in handover or roaming.

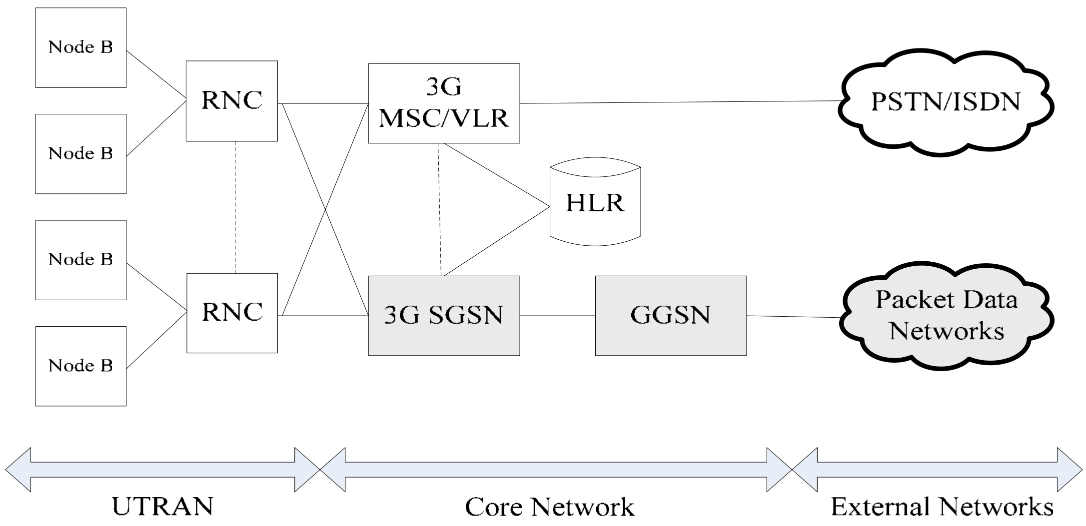


Figure1- 2. UMTS system architecture

1.2 Motivation

In order to support more applications and high transmission rates, 3G technologies have been developed. Since UMTS, one of 3G systems, is the extension of GSM system and provides backward compatible with GSM. UMTS has been deployed to overlap with the existing GSM system. In the early deployment, UMTS will be deployed mostly at urban area to save the initial capital cost. However, it needs to handover to GSM system in the border cells to ensure a seamless connection. To achieve that, there are three methods for handover as below:

a. Blind Handover

The mobile is not requested to perform measurements, and the target cell is chosen autonomously by the UTRAN. This speeds up the handover procedure but results in a lower handover success rate. For the inter-system handover procedure, it based on GSM measurement to guarantee a reliable handover success rate.

b. Dual Transceiver

The mobile is continuously transmitting and receiving on the carriers. In order to perform measurement, the mobile uses two transceivers for each carrier at the same time.

This method requires additional complexity and cost for the mobile. However, it might not enough to measure the third frequency band, like GSM 1800 band. The hardware architecture will extend endlessly.

c. Compressed Mode

In order to save the hardware complexity, the mobile will stop receiving on one carrier and receive temporarily another one. The UMTS proposes a mechanism called "Compressed Mode" [4], which interrupts the current connection to measure the carriers of other systems. The compressed mode is chosen both for reliable handover and less hardware overhead in advanced cellular system.

The UMTS specification has set up the regulars about the compressed mode. The specification suggests three methods to generate the transmission gap: (1) Reducing the spreading factor by two, (2) Puncturing, (3) Higher layer scheduling. These methods either speed up the transmission rate in physical layer or delay the data transmission in higher layer. During the compressed frame, more power is required to guarantee the quality of enhanced transmission rate. The increasing power will impact the performance, such as the capacity and coverage. If we want to measure more GSM carriers, the compressed mode is required frequently. However the more frequent compressed mode, the more impacts are suffered in performance. A capacity-based compressed mode is then proposed to tradeoff between the successfully handover and the performance.

1.3 Overview of UMTS

UMTS is a wideband Direct-Sequence Code Division Multiple Access (DS-CDMA) system. The user information bits are spread over a wide bandwidth by multiplying the user data from the spreading factor. The chip rate of 3.84 Mcps used leads to a carrier bandwidth of approximately 5 MHz. Each frame consists of 15 slots over 10 ms. The fast power control based on slot information is supported with 1.5 kHz frequency in both uplink and downlink. Then, some of the Radio Research Management (RRM) schemes [5] related to the compressed mode are introduced.

1.3.1 Spreading

DS-CDMA [5] spreads out the narrowband data by multiplying from the wideband spreading code. Each user has its own spreading code which is orthogonal to all other codes. Only the same spreading code data can be despreaded as the initial data from all other code data. Figure 1-3 depicts the basic operation of spreading and dispreading for a DS-CDMA system.

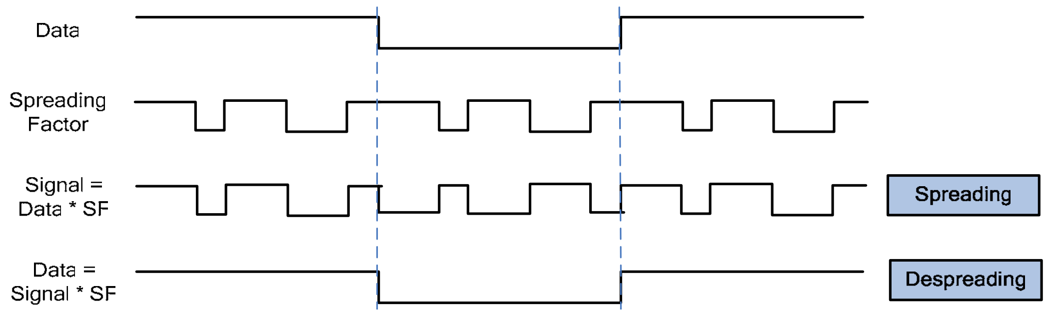


Figure1- 3. Spreading and dispreading in DS-CDMA

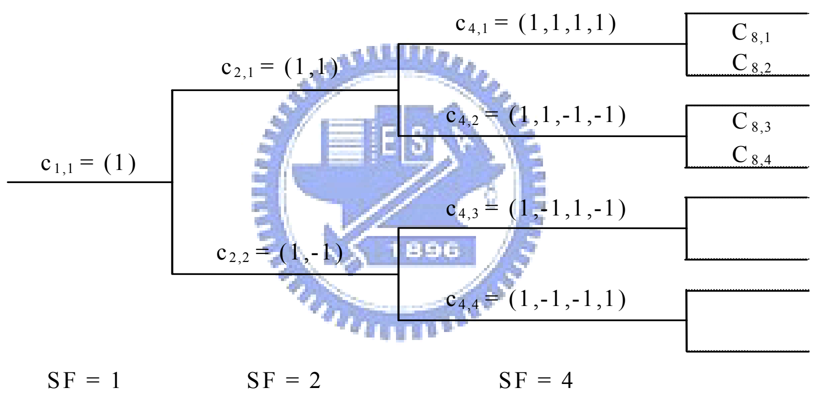


Figure1- 4. The OVSF code tree

CDMA users can transmit at higher data rates by using either multiple number of orthogonal constant spreading factor (OCSF) codes or orthogonal variable spreading factor (OVSF) code. The former supports higher data rates by assigning a multiple of OCSF codes to one call. It requires multiple transceiver units, thus resulting in increased hardware complexity. The later supports higher data rates by using lower OVSF. It may lead to a higher code blocking rate for higher data rate users. The OVSF code is suggested in UMTS standard [7] as depicted in Figure 1-4. The spreading code is derived from the lower spreading factor code. In IS-95 [8], the walsh code is generated from the former walsh code by equation 1-1.

$$W_{n+1} = \begin{bmatrix} W_n & W_n \\ W_n & -W_n \end{bmatrix} \tag{1-1}$$

Although the generation methods are different, the basic concepts and the generated codes are almost the same. The reason is only to avoid the IS-95 patents.

Based on the UMTS chip rate 3.84 Mcps, the channel rate equals to the chip rate divided by the spreading factor. The corresponding channel rates are listed in Table 1-1. When achieving higher data rate or reducing the spreading factor by two in the compressed mode, the shorter OVSF code is needed. How to assign them with avoiding the code blocking is another research. The existed methods are static code assignment [9], dynamic code assignment [10], region division OVSF code assignment [11], and hybrid OVSF code assignment [12]. All of these algorithms separate the longer code from the shorter code and do their best effort to enhance the code capacity.

Table1- 1. The channel rate with different spreading factors

DPDCH spreading factor	DPDCH channel rate (kbps)	User data rate with 1/2 coding rate (approx.)
256	15	7.5 kbps
128	30	15 kbps
64	60	30 kbps
32	120	60 kbps
16	240	120 kbps
8	480	240 kbps
4	960	480 kbps
4,with 6 parallel codes	5740	2.3 Mbps

1.3.2 Handover

When a mobile moves away from a base station, the signal level degrades. The weak signal strength can't support the original services, so there is a need to switch communication to another base station. Handover is the mechanism that supports the mobility of the user terminals. There are many important issues related to handover in Figure 1-5 [13]. These issues can divide into five parts: control, methodology, metric, parameter, and performance.

The handover control scenario consists of network-controlled, mobile-assisted, and mobile-controlled. The base station has the responsibility to do the measurement and decide the handover operation in network-controlled system. On the contrary, the mobile station has the responsibility in mobile-controlled system. In the mobile-assisted system, the mobile measures the strength of the base stations and then reports to the serving base station. The serving base station makes the final decision with the handover execution. Based on the system architecture, it chooses the proper handover control method.

The handover methodologies include hard, soft, and softer handover. “The mobile connects to the new base station after it breaks the origin base station” is called hard handover. The mobile has only one connection at the same time. The hard handover saves the usage of the radio resource but may cause a connection dropping. For guaranteeing the connection quality, the simultaneous connection is suggested. In soft handover there are multiple cells simultaneously support a call, and in softer handover there are multiple sectors simultaneously support the call. With the soft and softer handover, the call dropping rate is apparently reduced and the received signals have more confidence by the handover gain.

The handover algorithm divides into handover metrics and handover parameters. The algorithm can use pilot Ec/Io, RSCP, RSSI, SIR … etc. as the handover triggering metrics. The most popular criterion in CDMA system is the pilot Ec/Io, because it can reflect the influences of the noise. Other metrics are also used in the other radio interface systems. The handover parameters consist of hysteresis margin, dwell timer … etc. The hysteresis margin tolerates the rapidly fluctuation of the measured signal. The dwell timer avoids the ping-pong effect which means switch between two systems too frequently. At last, the handover performance reflects on call blocking, call dropping, delay … etc. According to these performances, the quality of the handover algorithm can be judged.

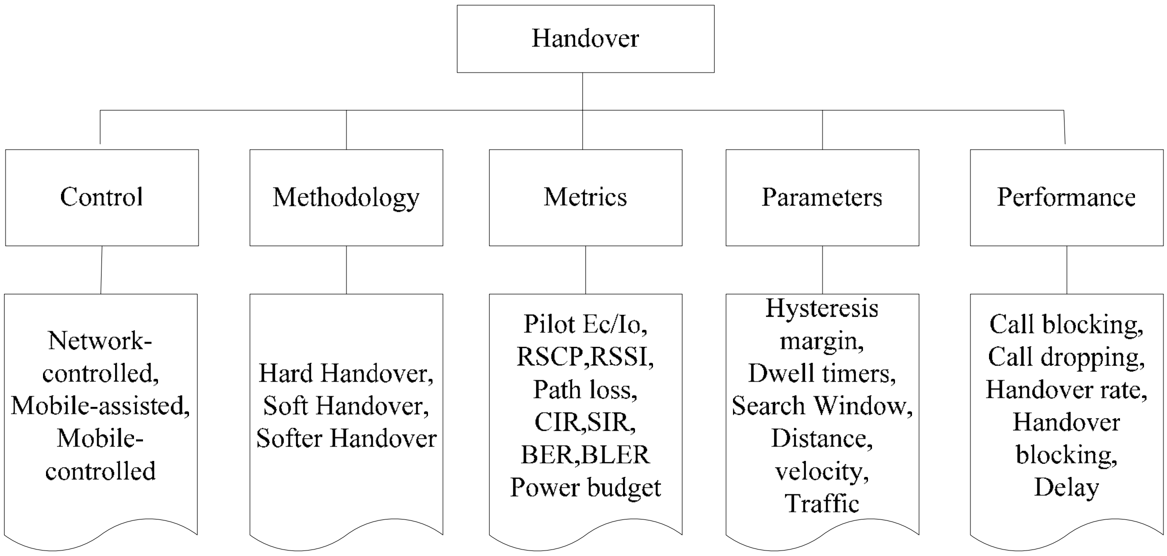


Figure1- 5. Important issues involved in the handoff mechanism

For example of UMTS, it is a mobile-controlled soft handover system. The soft handover uses pilot Ec/Io as the handover measurement quantity. The measured base stations are divided into below three sets:

Active Set:

The cells form a soft handover connection to the mobile user.

Monitored Set:

The cells are that the mobile user continuously measures, but whose pilot Ec/Io are not strong enough to be added into the active set.

Remaining Set:

Other cells are not in above two sets and whose pilot Ec/Io are usually weak.

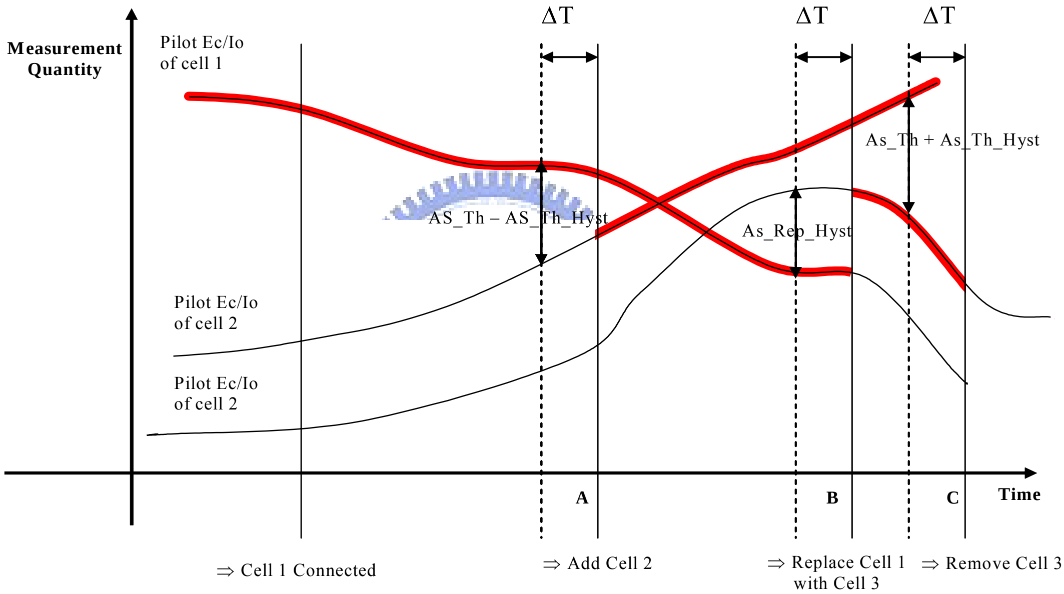


Figure1- 6. UMTS soft handover algorithm

The soft handover algorithm in standard [7] as described in Figure 1-6 is as follows:

- If the measured signal is greater than (the best measured signal in active set - As_Th + As_Th_Hyst) for a period of ΔT and the Active Set is not full, then the cell is added to the Active Set.
- If the measured signal is less than (the best measured signal in active set - As_Th - As_Th_Hyst) for a period of ΔT, then the cell is removed from the Active Set.
- If Active Set is full and the best measured cell in the Monitored Set is greater than (the worst measured cell in the Active Set + As_Rep_Hyst) for a period of ΔT, then the weakest cell in the Active Set is replaced by the strongest Monitored Set.

Where:

AS_Th: Threshold for macro diversity (reporting range);

AS_Th_Hyst is the hysteresis for the above threshold.

AS_Rep_Hyst is the replacement hysteresis.

ΔT: Time to Trigger.

The soft handover algorithm can be easily modified the parameter or add more condition to enhance the performance. The thresholds are modified and the performance is evaluated in [14]. With the soft handover, the seamless connection can be guaranteed and the connection quality also can be maintained.

1.3.3 Power Control

Power control procedures determine the power levels for transmission to and from the terminal. The effective power control can enhance the battery life and avoid the near-far problem which means the signal of near terminal may cover the far ones. If the terminal moves closer to the serving base station, the system may reduce the power levels in order to reduce interference to other calls. After processing power control, all received power levels are desired to be equal.

The UMTS power control procedure [15] is depicted in Figure 1-7. When the User Equipment (UE) initially accesses to the network in uplink, it executes the open loop power control to achieve the target power level. After the connection has been set up, the close loop power control is then executed to adjust the power level. The close loop power control includes of inner loop and outer loop power control. The outer loop power control set the target signal to interference ration, SIR_{target} according to the connection quality (in terms of FER). With the basis of the SIR_{target}, the inner loop power control increases or reduces the power level.

The inner loop power control increases the power level if SIR is larger then SIR_{target} and decreases the power level otherwise. There are two alternative algorithms which the base station can instruct the mobile for modifying the power level. “Algorithm 1” is designed for use when the mobile speed is sufficiently low for inner-loop power control to act against the fading. The mobile simply increases and reduced power by 1 dB. “Algorithm 2” is designed to emulate the effect of using a step size smaller than 1 dB and is effective to avoid the rapid short-term fluctuations. The mobile increases and reduced power by 1dB only if the power control commands are continuously consistence in 5 times. Algorithm 2 gives better performance than algorithm 1 for stationary users and for high-speed users.

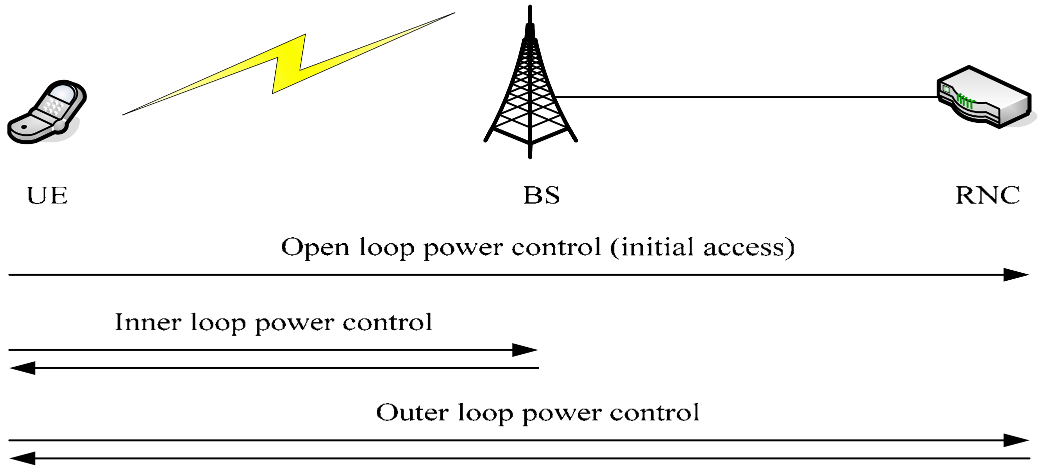


Figure1- 7. UMTS power control procedure

Unlike the 1.5 kHz fast inner loop power control, the frequency of the outer loop power control is typically 10-100Hz. While the frame error appears and the received data quality is less than the expected value, the network increases SIR_{target} to enhance the signal strength. Otherwise the network decreases SIR_{target} to reduce the interference. The whole close loop power control scheme is depicted in Figure 1-8. With this power control procedure, the received signal strength is always allocated at the proper range and the connection quality can then be satisfied.

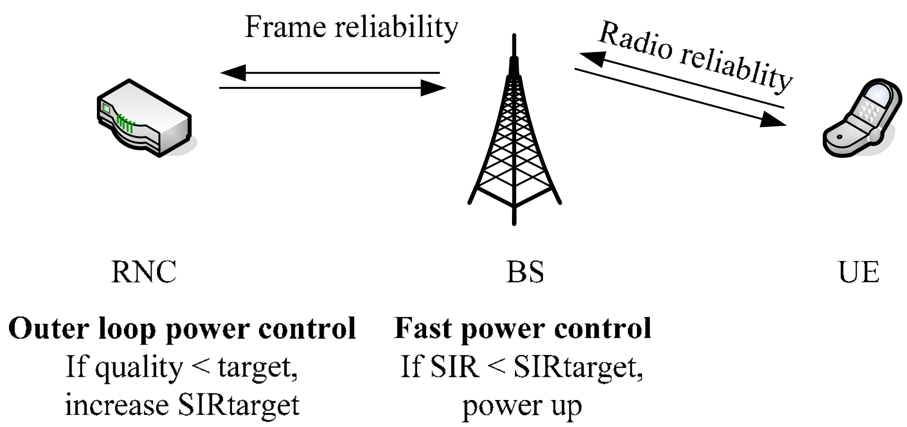


Figure1- 8. UMTS close loop power control scheme

1.4 Overview of Quality of Service

The multimedia services and the file download services are become more and more popular. However the different services have different constrains. In order to satisfy all users with different services, the Quality of Service (QoS) needs to be defined. The classic QoS attributes include the following kinds:

- The traffic characteristics specified (in terms of bandwidth):

  Ex: The peak rate, the minimum acceptable rate, the average rate, the maximum burst size …
- The reliability requirements of the connection:

  Ex: Bit Error Rate (BER), Frame Error Rate (FER), the maximum loss rate …
- The delay requirements:

  Ex: The maximum tolerated delay, the maximum tolerated jitter …

In [16], the author also proposed two advanced QoS attributes for the wireless communication. The Seamless Service Descriptor (SSD) describes the type of seamless service the user is requesting, and the Service Degradation Descriptor (SDD) describes how much the user is willing to get a degraded service. Both of these two attributes are according to the user profile. The user profile may determine by the bill paying.

Table1- 2. UMTS QoS classes

Traffic class	Fundamental characteristic	Application
Conversational class	Conservative real-time No ARQ (retransmission) High sensitivity to delay and jitter	Voice, video-telephony
Streaming class	Streaming real-time No ARQ High sensitivity to jitter Medium sensitivity to delay	Streaming multimedia
Interactive class	Interactive best effort ARQ High sensitivity to round trip delay High sensitivity to BER Low sensitivity to delay	Web browsing, network games
Background class	Background best effort ARQ High sensitivity to BER No delay sensitivity	Background download, e-mail

In UMTS standard, four traffic classes have been identified [17]. Two real time services (conversational, streaming) and two best effort service (interactive, background) are suggested. The fundamental characteristic and the application of each class are described in Table 1-2. The real time services emphasize on the delay requirements and the best effort services consider about the reliability requirements. According to the different service class, the system gives the different research and creates the maximum satisfaction.

1.5 Thesis Organization

The organization of this thesis is described as follows: Chapter 2 introduces the detail operations of the compressed mode mechanism. The physical behaviors are studied and the performance impacts are also included. In chapter 3, the problem of existed compressed mode is addressed and the enhanced control algorithm is also proposed. Chapter 4 discusses the simulation platform capability. In chapter 5, some experimental results based on the proposed compressed mode algorithm are shown. The performances of power consumption and measurement efficiency are illustrated to verify the proposed algorithm. Finally, chapter 6 gives the contributions and the advanced future works.

Chapter2

Overview of the Compressed Mode

This chapter introduces the detail operations of the compressed mode in the physical layer. Based on this architecture, the metrics of the GSM measurement are also addressed. Finally, the performance impacts caused by the compressed mode are discussed.

2.1 The Compressed Mode Architecture

In wireless communication systems, in order to keep a seamless connection, the handover scheme is essential. From 2^{nd} generation, the mobile assisted handover is applied, and the mobile has the responsibility to measure other base stations' power strength. In GSM/GPRS which are time division multiple access (TDMA) systems, the mobiles can use the slots non-transmitting to measure other carriers. In contrast to CDMA/UMTS which are code division multiple access (CDMA) systems, the mobiles transmit continuously. The measurement of other carriers becomes a problem. In UMTS, the compressed mode, with variable transmission gaps, measures the other carriers' strength and is depicted in Figure 2-1.

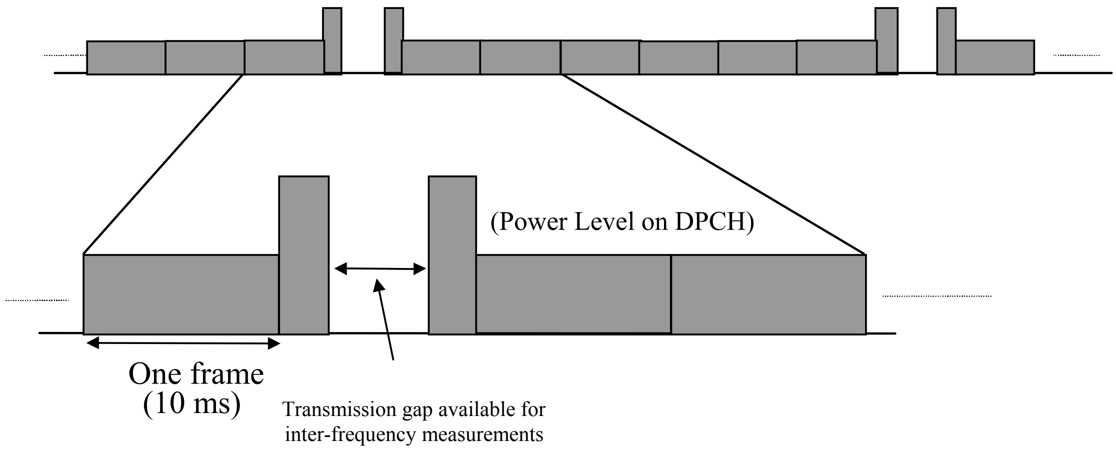


Figure2- 1. The compressed mode transmission

For the inter-system handover in UMTS, the compressed mode is operated as follows: First, the mobiles in UMTS cells need to measure the carrier strength of other systems during the transmission gap. Then, the mobiles acquire the control channel message of the measured system based on the associated carrier strength. When the information of the control channel has been collected, the handover operation will then be performed. The compressed frames can occur periodically or requested on demand. The frequency and type of the compressed mode depends on the measurement requirements, and is decided by the network.

2.1.1 The Gap Generation Methods

The UMTS implements the compressed mode to execute the inter-system handover. In UMTS standard, following three methods are suggested to prevent the data lost [4]:

- Reducing the spreading factor by two

  The first method is to double the data rate by reducing the spreading factor by two. The time saved from accelerating the data transmission rate is used for the transmission gap. This method is not supported for SF=4, because 4 is the minimum value of allowed spreading factor. Besides, more power is also required for supporting the higher transmission rate.
- Puncturing

  The second method is to puncture the redundancy bits made by channel coding in the transport channel to allow the system to generate the gaps. This method modifies only the channel coding rate but keeps the existing spreading factor and bit rate unchanged. Due to the limitation of the physical channel format and the channel coding rate, this method applies to downlink transmission and short transmission gap only.
- Higher layer scheduling

  Besides of increasing the speed of the physical layer transmission, the data can be processed at the higher layer for non-real time data services or voice services with enough silence time. Since the number of bits can be rescheduled, a gap can then be generated for a non-schedule period. This method does not affect the physical transmission rate so it has the least impacts on the performance but is only suitable for non-real time services.

Due to the limitations of implementing the "puncturing" and "higher layer scheduling" schemes, in this thesis "reducing the spreading factor by two" is chosen for the rest studies to satisfy the entire transmission scenarios.

Beside these three methods, the system can easily give other alternatives to increase the transmission rate and then leave time for transmission gap. For example, the formula of the transmission bit rate, R_b, is defined as equation (2-1) in [18]:

$$R_b = (1-\gamma)\cdot N \cdot \frac{R_c \cdot R \cdot M}{SF} \tag{2-1}$$

where γ (the compressed mode ratio) is defined as the fraction of the frame left idle during the compressed mode, N is the number of parallel physical CDMA code channels used, R_c is the chip-rate, R is the channel encoding rate, M is the modulation order, and SF is the spreading factor.

Base on this equation, the transmission bit can easily increased by modify the parameter in the left part. Then the system can propose four methods to generate the transmission gap [18]:

a. Variable Spreading Factor Compressed Mode (VCF-CM)

b. Code-Rate Increased Compressed Mode (CRI-CM)

c. Multi-Code Compressed Mode (MC-CM)

d. Higher Order Modulation Compressed Mode (HOM-CM)

However, some methods, like method c and method d, change the radio interface architecture and then increase the hardware complexity. Other methods, like method b and method d, compact the signal constellation and then decrease the reliability of transmitted data. All these methods will be chosen only if the QoS constraints are satisfied.

2.1.2 The Transmission Gap Position and the Frame Structure

In UMTS, there are 15 slots in one frame and each frame spans 10 ms. The compressed mode has many types of Transmission Gap Length (TGL) and the TGL formats are equal to only 3, 4, 5, 7, 10, 14 slots [4]. The gap position is depicted as Figure 2-2, where N_{first} is the first slot of the transmission gaps and N_{last} is the last slot of the gaps. In order to ensure the power control successfully, there are at least 8 slots transmitted in each radio frame. The transmission gap should be divided into two types, the single frame method and double frame method. Thus N_{first} and TGL must be chosen to correspond with this criterion.

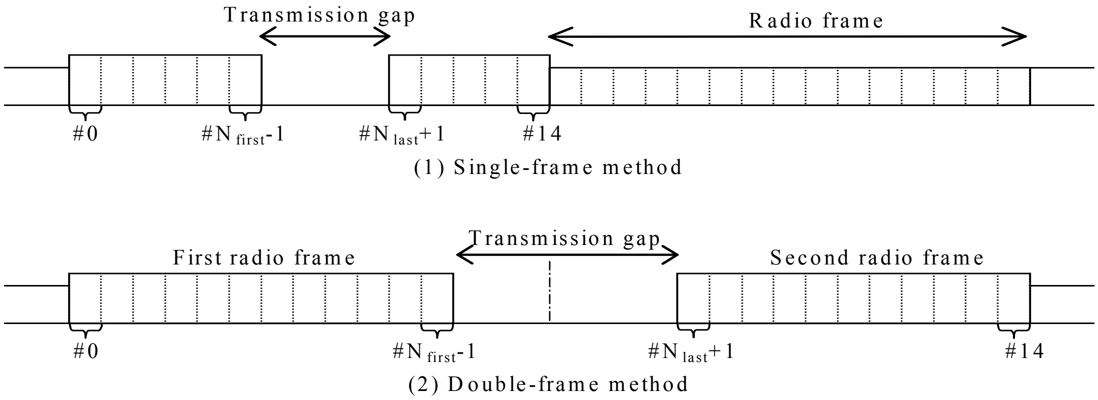


Figure2- 2. Transmission gap position

The framing structures of the uplink and downlink are depicted in Figure 2-3 and Figure 2-4. In uplink, the Dedicated Packet Data CHannel (DPDCH) and the Dedicated Packet Control CHannel (DPCCH) are transmitted separately. The details of frame formats are listed in Table 2-1 and Table 2-2 [19]. In the uplink DPDCH fields, the data is transmitted with BPSK and the network modifies the spreading factor to change the transmission rate. In the uplink DPDCH fields, the data rate keeps the minimum rate, 15 kbps, to guarantee the reliable of the control signals. There are two possible compressed slot formats labeled as A and B in DPCCH. Label A has smaller gaps for the compressed mode and its number of transport format combination indicator (N_{TFCI}) bits is also less for easy combination.

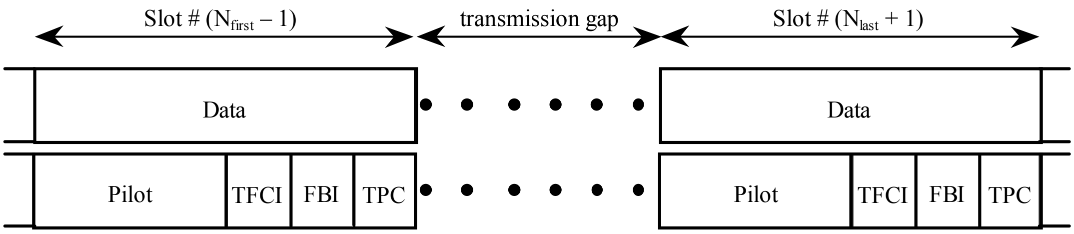


Figure2- 3. Frame structure in uplink compressed transmission

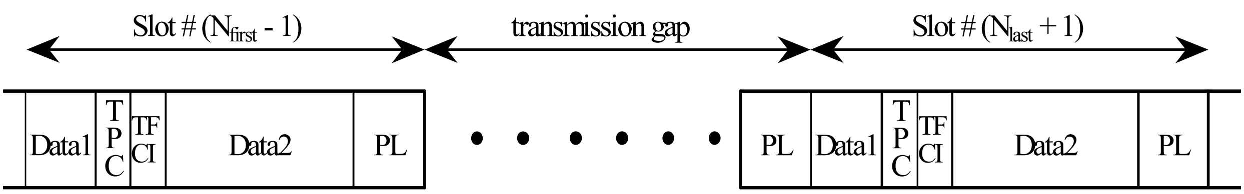
Table2- 1. Uplink DPDCH fields

Slot Format #i	Channel Bit Rate (kbps)	Channel Symbol Rate (ksps)	SF	Bits/ Frame	Bits/ Slot	N_{data}
0	15	15	256	150	10	10
1	30	30	128	300	20	20
2	60	60	64	600	40	40
3	120	120	32	1200	80	80
4	240	240	16	2400	160	160
5	480	480	8	4800	320	320
6	960	960	4	9600	640	640

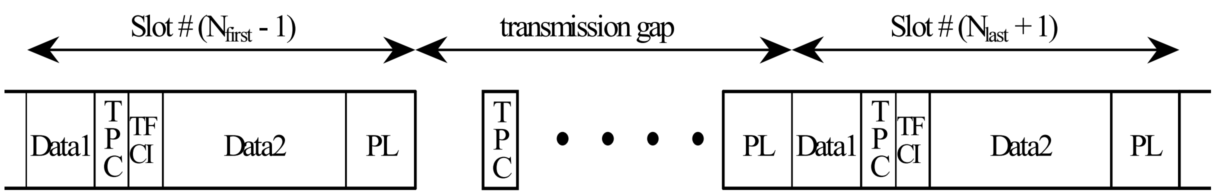
Table2- 2. Uplink DPCCH fields

Slot Format #i	Channel Bit Rate (kbps)	Channel Symbol Rate (ksps)	SF	Bits/ Frame	Bits/ Slot	N_{pilot}	N_{TPC}	N_{TFCI}	N_{FBI}	Transmitted slots per radio frame
0	15	15	256	150	10	6	2	2	0	15
0A	15	15	256	150	10	5	2	3	0	10-14
0B	15	15	256	150	10	4	2	4	0	8-9
1	15	15	256	150	10	8	2	0	0	8-15
2	15	15	256	150	10	5	2	2	1	15
2A	15	15	256	150	10	4	2	3	1	10-14
2B	15	15	256	150	10	3	2	4	1	8-9
3	15	15	256	150	10	7	2	0	1	8-15
4	15	15	256	150	10	6	2	0	2	8-15
5	15	15	256	150	10	5	1	2	2	15
5A	15	15	256	150	10	4	1	3	2	10-14
5B	15	15	256	150	10	3	1	4	2	8-9

In downlink transmission, the data channel and the control channel are interlacing into Dedicated Packet CHannel (DPCH). There are two types of frame structure labeled as A and B. In type A, the Pilot (PL) field of the last slot in the transmission gap is transmitted for synchronization. In type B, an additional Transmit Power Control (TPC) field of the first slot in the transmission gap is transmitted for optimizing the power control. The detail numbers of downlink DPCH bits are listed in Table 2-3 [19]. The data in downlink is transmitted with QPSK and the network modifies the spreading factor to change rate. There are also two possible compressed slot formats labeled as A and B. The slot format B shall be used by "reducing the spreading factor by two", and the slot format A shall be used by "puncturing" or "higher layer scheduling".



(a) Type A: maximize the transmission gap length



(b) Type B: optimized the power control

Figure2- 4. Frame structure types in downlink compressed transmission

Table2- 3. Downlink DPCH fields

Slot Format #i	Channel Bit Rate (kbps)	Channel Symbol Rate (ksps)	SF	Bits/ Slot	DPDCH Bits/Slot		DPCCH Bits/Slot			Transmitted slots per radio frame, N_{Tr}
					N_{Data1}	N_{Data2}	N_{TPC}	N_{TFCI}	N_{Pilot}	
0	15	7.5	512	10	0	4	2	0	4	15
0A	15	7.5	512	10	0	4	2	0	4	8-14
0B	30	15	256	20	0	8	4	0	8	8-14
1	15	7.5	512	10	0	2	2	2	4	15
1B	30	15	256	20	0	4	4	4	8	8-14
2	30	15	256	20	2	14	2	0	2	15
2A	30	15	256	20	2	14	2	0	2	8-14
2B	60	30	128	40	4	28	4	0	4	8-14
3	30	15	256	20	2	12	2	2	2	15
3A	30	15	256	20	2	10	2	4	2	8-14
3B	60	30	128	40	4	24	4	4	4	8-14
4	30	15	256	20	2	12	2	0	4	15
4A	30	15	256	20	2	12	2	0	4	8-14
4B	60	30	128	40	4	24	4	0	8	8-14
5	30	15	256	20	2	10	2	2	4	15
5A	30	15	256	20	2	8	2	4	4	8-14
5B	60	30	128	40	4	20	4	4	8	8-14
6	30	15	256	20	2	8	2	0	8	15
6A	30	15	256	20	2	8	2	0	8	8-14
6B	60	30	128	40	4	16	4	0	16	8-14
7	30	15	256	20	2	6	2	2	8	15
7A	30	15	256	20	2	4	2	4	8	8-14
7B	60	30	128	40	4	12	4	4	16	8-14
8	60	30	128	40	6	28	2	0	4	15
8A	60	30	128	40	6	28	2	0	4	8-14
8B	120	60	64	80	12	56	4	0	8	8-14
9	60	30	128	40	6	26	2	2	4	15
9A	60	30	128	40	6	24	2	4	4	8-14
9B	120	60	64	80	12	52	4	4	8	8-14
10	60	30	128	40	6	24	2	0	8	15
10A	60	30	128	40	6	24	2	0	8	8-14
10B	120	60	64	80	12	48	4	0	16	8-14
11	60	30	128	40	6	22	2	2	8	15
11A	60	30	128	40	6	20	2	4	8	8-14
11B	120	60	64	80	12	44	4	4	16	8-14
12	120	60	64	80	12	48	4	8*	8	15
12A	120	60	64	80	12	40	4	16*	8	8-14
12B	240	120	32	160	24	96	8	16*	16	8-14
13	240	120	32	160	28	112	4	8*	8	15
13A	240	120	32	160	28	104	4	16*	8	8-14
13B	480	240	16	320	56	224	8	16*	16	8-14
14	480	240	16	320	56	232	8	8*	16	15
14A	480	240	16	320	56	224	8	16*	16	8-14
14B	960	480	8	640	112	464	16	16*	32	8-14
15	960	480	8	640	120	488	8	8*	16	15
15A	960	480	8	640	120	480	8	16*	16	8-14
15B	1920	960	4	1280	240	976	16	16*	32	8-14
16	1920	960	4	1280	248	1000	8	8*	16	15
16A	1920	960	4	1280	248	992	8	16*	16	8-14

From the above discussions, the related parameters are summarized in Table 2-4. It lists the difference transmission time reduction method between uplink and downlink, and the related transmission gaps and frame formats.

Table2- 4. Parameters for combined UL/DL compressed mode

TGL	DL Frame Type	Spreading Factor	Idle length [ms]	Transmission time Reduction method	Idle frame Combining
3	A or B	DL: 512 – 4 UL: 256 – 4	1.47 – 1.73	DL: Puncturing, Spreading factor division by 2 or Higher layer scheduling UL: Spreading factor division by 2 or Higher layer scheduling	(S) (D) =(1,2) or (2,1)
4			2.13 – 2.39		(S) (D) =(1,3), (2,2) or (3,1)
5			2.80 – 3.06		(S) (D) = (1,4), (2,3), (3, 2) or (4,1)
7			4.13 – 4.39		(S) (D)=(1,6), (2,5), (3,4), (4,3), (5,2) or (6,1)
10			6.13 – 6.39		(D)=(3,7), (4,6), (5,5), (6,4) or (7,3)
14			8.80 – 9.06		(D) =(7,7)

2.2.3 The Parameters of the Compressed Mode

In response to request the compressed mode from higher layers, the UTRAN shall signal to the UE the compressed mode parameters [20]. Parts of parameters indicate the position of the transmission gap pattern and are depicted in Figure 2-5. The position parameters include Transmission Gap Starting Slot Number (TGSN), Transmission Gap Length (TGL), Transmission Gap start Distance (TGD), Transmission Gap Pattern Length (TGPL), Transmission Gap Pattern Repetition Count (TGPRC), and Transmission Gap Connection Frame Number (TGCFN). Based on these parameters, the connection generates two transmission gap patterns to measure carriers of other systems.

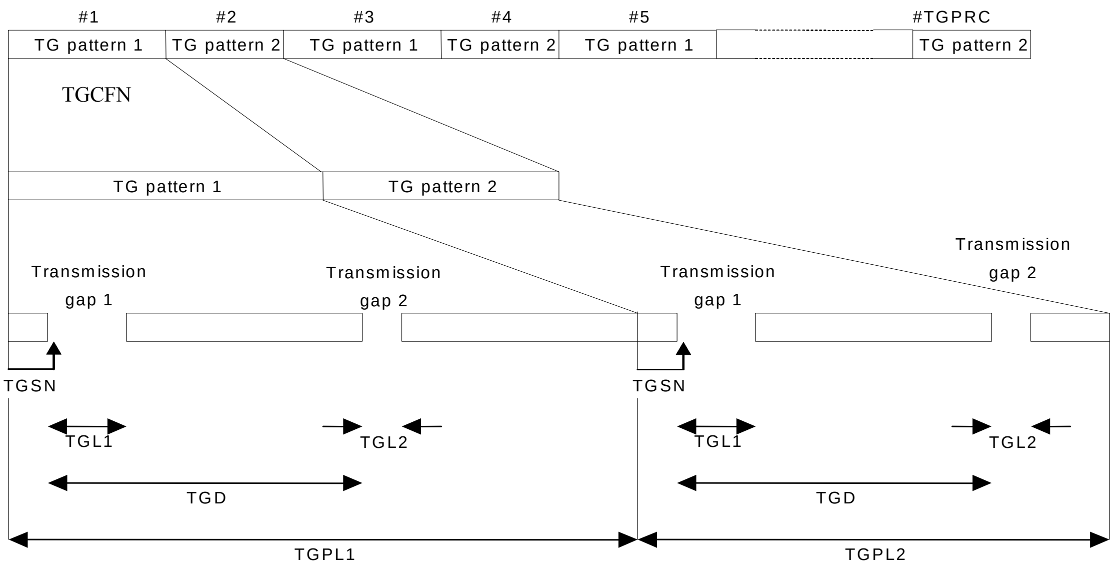


Figure2- 5. Illustration of the compressed mode pattern parameters

Besides the parameters defining the positions of transmission gaps, the physical behaviors of the compressed mode also characterized by following parameters:

- UL/DL compressed mode selection:

  This parameter specifies whether the compressed mode is used in uplink only, downlink only or both uplink and downlink. In most cases compressed frames are used in the downlink. If they are necessary in downlink and uplink, they must need time-aligned with some time offset.

- UL/DL compressed mode method:

  The methods for generating the uplink or downlink compressed mode gap are puncturing, spreading factor division by two or higher layer scheduling. The method is decided by the requested services and channel condition.

- Downlink frame type:

  This parameter defines if frame structure type 'A' or 'B' shall be used in downlink compressed mode. The frame structure type A maximizes the transmission gap and the type B is optimized for power control.

- Scrambling code change:

  This parameter indicates whether the alternative scrambling code is used for the compressed mode method 'reducing the spreading factor by two'.

- RPP:

  Recovery Period Power control mode specifies the uplink power control algorithm applied during recovery period after each transmission gap in the compressed mode. The recovery period enlarges the control step sizes and accelerates the power control.

- ITP:

  Initial Transmit Power mode selects the uplink power control method to calculate the initial power after the gap. The initial transmit power control resumes the power to initial level immediately.

Higher layers will base on those parameters to execute the compressed mode. It should ensure that the compressed mode gaps do not overlap and the performance is satisfied the QoS constraints.

2.2 Power Control on the Compressed Mode

Since there is no power control during the transmission gaps, it needs higher power to maintain the connection quality (BER, FER, etc.). In the compressed mode, the power level is increased and the power control scheme is different from the normal situation [15].

2.2.1 Uplink Power Control

The basic formula for uplink power control is in equation (2-2), where ΔDPCCH is the change of power level in DPCCH, ΔTPC is the step size per change, and TPC_cmd is the power control command to decide whether to increase or decrease the transmitted power.

$$\Delta DPCCH = \Delta TPC \times TPC\_cmd \tag{2-2}$$

If the estimated Signal to Interference Ratio SIR_{est} is greater than the preset target SIR SIR_{target}, then TPC_cmd is equal to -1 and the power level is decreased toward SIR_{target} on next slot. On the contrary, if SIR_{est} is less than SIR_{target}, then TPC_cmd is equal to 1 and the power level is increased toward SIR_{target}.

During the compressed mode, the power level might need to be increased to ensure the quality, and the target SIR increases as SIR_{cm_target} depicted in equation (2-3). The frame which exists of transmission gap and the next frame should increase DeltaSIR and DeltaSIRafter for connection quality. From Table 2-2, it can be observed that pilot bits in uplink compressed mode will be decreased, and the synchronization ability will be degraded. The mobile should increase addition power level to achieve synchronized.

$$SIR_{cm\_target} = SIR_{target} + \Delta_{PILOT} + DeltaSIR + DeltaSIRafter \tag{2-3}$$

In additional after the transmission gap, UMTS proposes two optional mechanisms as the power resume mode and the power recovery mode to accelerate power control. The power resume mode sets the mobile initial transmit power after a transmission gap. This can take one of two values: either the same power as immediately before the gap, or a value equal to the approximated average of the transmit power by the recursive function listed in equation (2-4). The power resume mode resumes the initial power and restarts the power control.

$\delta_i = 0.9375\delta_{i-1} - 0.96875 \text{ x (Most\_Recent\_power\_change)}$ (2-4)

$\delta_{i-1} = \delta_i$

The second mechanism provided to optimize power control after the transmission gaps is the power recovery mode. The power recovery mode controls the power control step size and algorithm as listed in Table 2-5 for a number of slots after each transmission gap. This mode replaces the slower power control algorithm 2 to the faster algorithm 1, and enlarges the step size. By operating power recovery mode, the mobile can accelerate the power control and speed up to acquire the correct power level.

Table2- 5. Power control in power recovery mode

Outside Recovery Period	In Recovery Period	
	Algorithm	Step Size (dB)
Alg. 1, 1dB step	1	2
Alg. 1, 2dB step	1	3
Alg. 2, 1dB step	1	1

2.2.2 Downlink Power Control

The basic formula for downlink power control is listed in equation (2-5), where $P_{TPC}(k)$ is the k^{th} power adjustment due to the inner loop power control, and $P_{bal}(k)$ is a correction according to the downlink power control procedure for balancing radio link powers towards a common reference power.

$P(k) = P(k - 1) + P_{TPC}(k) + P_{bal}(k)$ (2-5)

During the compressed mode, the base station also needs to increase power level by adding a term $P_{SIR}(k)$ for ensuring the quality and is modified to equation (2-6). This added term can be formulated as equation (2-7). The frame which exists of transmission gap and the following frame should increase DeltaSIR and DeltaSIRafter. With different compressed method, different ΔP_i_compression values have been applied as Table 2-7.

$P(k) = P(k - 1) + P_{TPC}(k) + P_{SIR}(k) + P_{bal}(k)$ (2-6)

$P_{SIR}(k) = \Delta(\Delta P_i\text{\_compression} + \text{DeltaSIR} + \text{DeltaSIRafter})$ (2-7)

When the reducing the spreading factor by two method has been applied, the most data rate is increased. And thus the largest power, 3 dB, is increased. When the puncturing method is applied, the increased power is according to the transmission gap length. When the higher layer scheduling method is applied, the data rate maintains the same value. And thus power rising is not needed.

Table2- 6. ΔP_i_compression value with different methods

ΔP_i_compression	**Method**
3 dB	Reducing the spreading factor by two
10 log (15*F_i / (15*F_i - TGL_i)) dB	Puncturing
0 dB	High layer scheduling or Non-compressed frame

Finally, the power increasing scenario is simply summarized in Figure 2-6, where the transmission rate could be increased to compensate the “no transmission” during the gaps. With higher transmission rate and no power control during the transmission gaps, the system should increase the transmission power to overcome unpredictable channel variance at the compressed frame and next frame by DeltaSIR and DeltaSIRafter. Besides, the mobile increases the power to compensate the reduced pilot bits used for synchronization in the uplink as ΔPILOT . In the downlink, a base station also increases the corresponding power by ΔPi_compression to ensure the quality of increased data rate.

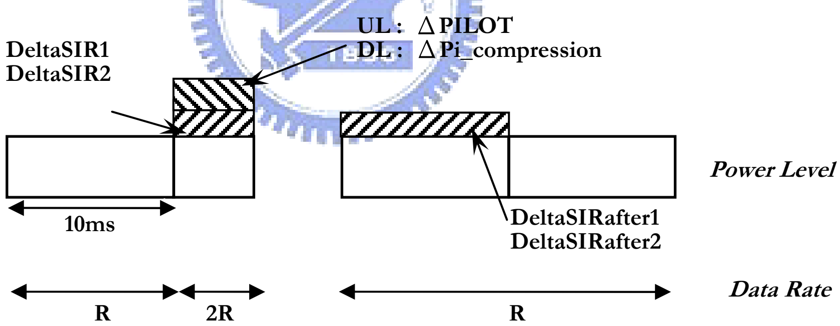


Figure2- 6. Power increasing scenario in the Compressed Mode

2.3 Measurements of GSM carriers

The system can measure FDD, TDD, or GSM carriers during transmission gap. Now, the following will talk about the GSM measurements with three steps [21]: GSM carrier Received Signal Strength Indicator (RSSI) measurement, Initial Base Station Identity codes (BSIC) identification, and BSIC re-confirmation.

➢ GSM carrier RSSI measurement

A UE supports GSM measurements using the compressed mode shall meet the minimum number of GSM RSSI carrier measurements specified in Table 2-7. The UE measures parts of the samples of carriers in the first transmission gap, and measures the remaining samples in the next gaps until complete measuring all carriers. The gaps should synchronize to transmitting frames in GSM, such as Frequency Correlation CHannel (FCCH) or Synchronization CHannel (SCH), to ensure detecting the GSM carrier RSSI successfully. For example [22]: If the pattern with 7 slot gap every 3rd frame is applied, and assuming the maximum 32 carriers, which need 3 samples for each carrier in GSM neighbor list. The UE needs (3*32)/6 * 30ms = 480ms to complete the measurement. Note that 480ms is equal to the time that one Slow Associated Contorl Channel (SACCH) message has been transmitted in GSM. The timing of measurement interval is almost the same.

Table2- 7. The gap length and GSM carrier RSSI measurement

TGL	Number of GSM carrier RSSI samples in each gap.
3	1
4	2
5	3
7	6
10	10
14	15

- Initial BSIC identification

The initial BSIC identification includes searching for the BSIC, which is transmitted in SCH, and decoding the BSIC for the first time when there is no knowledge about the relative timing between the FDD and GSM cell. The UE shall trigger the initial BSIC identification within the available transmission gap in Table 2-8. The UE shall use the last available GSM carrier RSSI measurement results and arrange GSM cells in signal strength order for performing BSIC identification. For GSM cells that are requested to decode the BSIC with 8 strongest carriers. If the UE has not successfully decoded the BSIC within a preset abort time, the UE shall abort the BSIC identification attempt and searching the second strong carrier.

Table2- 8. The gap length and maximum time difference for BSIC verification

Gap length [slots]	Maximum time difference [μs]
5	± 500
7	± 1200
10	± 2200
14	± 3500

- BSIC re-confirmation

  The BSIC re-confirmation is tracking and decoding the BSIC of a GSM cell after initial BSIC identification is performed. The UE shall trigger the BSIC re-confirmation within the available transmission gap. If the UE fails to decode the BSIC after two successive attempts or if the UE has not been able to re-confirm the BSIC within a preset abort time, the UE shall abort the BSIC re-confirmation attempts and then move to the initial BSIC identification procedure again.

2.4 The Performance Impacts

Although the compressed mode could help on the inter-system handover, some system performance besides of the signal quality (such as Bit Error Rate or Frame Error Rate) will also be impacted:

- Coverage

  Because more power is needed for the same QoS constraint during the compressed mode, the maximum path loss budget is reduced. As a result, the distance that the mobile can transmit is reduced, thus the uplink coverage is reduced. In [5], the path loss will be reduced by 2.3dB if compressed frame is operated every second frame.

- Capacity

  Since the power control cannot work during compressed frame, the transmitted data needs higher E_b/N_0 requirement to maintain the quality. The capacity will be decreased by the higher E_b/N_0 requirement. The capacity can be reduced by about 2% even only 10% of users are operating the compressed mode every third frame [5].

- Code Shortage Problem

  When the “reducing the spreading factor by two” method is applied, the blocked code is twice than before and the available code space is reduced. There might not have enough codes to support all active users especially for high data rate users (low spreading factor). There are three proposed solutions for the code shortage problem: 1. Using Non-orthogonal scrambling codes. 2. Avoid the transmission gaps from different mobiles overlap in one frame, the spreading factor would be used in fair distribution [23]. 3. Base station assigns a dynamic common channel to a mobile station performing inter-frequency/system handover [24].

Here, the challenges are how to find a robust mechanism to get the best efficiency and performance by allocating the transmission gaps in various fading and the relative uplink/downlink power control before and after the transmission gaps for different fading condition. It is critical to have an adaptive compressed mode operation to minimize the impacts on system performance.

Chapter 3
Capacity-Based Compressed Mode

The performance impacts of the compressed mode are studied. To reduce the impacts and maintain the compressed mode efficiency, a capacity-based compressed mode is proposed in this chapter. The concepts and the implement of the proposed control algorithm are discussed here.

3.1 The Prior Works

So far, many articles discuss the compressed mode performance. However, they mostly modify compressed mode algorithms to enhance the border-cell handover success rate without concerning the potential impacts caused by the compressed mode. Besides of three compressed mode methods suggested by 3GPP standard [4], Gustafsson et al. [18] provide the formula of the transmission bit rate and relative parameters to generate the transmission gap. Moreover, the issues of triggering criteria for the compressed mode are researched. Ying et al. [25] compare the periodic and event-triggered compressed mode. With extra overhead cost, the period-triggering algorithm has higher handover success rate. Zhang [26] considers the base line quality and suggests that the pilot Ec/Io is suitable for high traffic load and the pilot received signal code power (RSCP) is suitable for low traffic load. The combined pilot Ec/Io and pilot RSCP triggering method is then proposed to guarantee good system performance under all traffic loads. The relationship of the compressed mode gap pattern and GSM carriers measurement are considered in [22][27]. The required handover time for different transmission gap patterns is studied.

During each compressed frame, more power is suggested to guarantee the quality of increased transmission rate. The interference caused by the increasing power will impact the performance in either capacity or coverage. Holma and Toskala [5] quantify the degradation of the capacity and coverage. Some discussions in 3GPP TSGR4 meetings also address the impact scenarios [28-30]. Although the above articles point out the performance impacts, no proper solution is proposed. To resolve that, a capacity-based compressed mode is proposed to balance the tradeoff between the handover success rate and the capacity.

If the mobiles in the border of UMTS cells want to hand down to GSM cells, it uses the compressed mode to measure GSM carriers. However the GSM measurement is often not enough in current implementation. The critical problem is that measure at wrong time when GSM doesn't transmit signals. With the wrong time measurement, not only measure no signal but also waste power without any profit. It prefers that all mobiles measure at the right time simultaneously and then use the proposed capacity-based compressed mode to limit the interference level. It benefits both effective measurements and performance maintenance.

3.2 The Concept of the Capacity-Based Compressed Mode

The increasing power is to compensate the influence of the transmission gaps, but it also impacts the performance on capacity or coverage. In this thesis, the fixed cell size is calculated by the maximum path loss, so only the capacity impacts are considered. To calculate the uplink capacity, the required bit energy to interference ratio (E_b/I_0) can be computed in equation (3-1). For simplicity, the same traffic services and the same received power at the base station are assumed for all mobiles.

$$\frac{E_b}{I_o} \approx \frac{S \cdot PG}{FN_{th}W + \alpha[(1+\beta)(N-1)S + N_{CM}\Delta S]} \tag{3-1}$$

where $\boldsymbol{E_b}$ is the received bit energy, $\boldsymbol{I_o}$ is the total interference, $\boldsymbol{S}$ is the received power at the base station, $\boldsymbol{PG}$ is the processing gain, $\boldsymbol{F}$ is the noise figure, $\boldsymbol{N_{th}}$ is the thermal noise power density, $\boldsymbol{W}$ is the transmission bandwidth, α is the voice activity, β is the adjacent cell interference factor, $\boldsymbol{N}$ is the number of users, $\boldsymbol{N_{CM}}$ is the number of users in the compressed mode, and $\boldsymbol{\Delta S}$ is the average increasing power for the compressed mode. The capacity, $\boldsymbol{Nc}$, can then be calculated in equation (3-2), where $\left(\frac{E_b}{I_o}\right)_{target}$ is the target bit energy to interference ratio. The capacity will be deducted by $\left(\frac{N_{CM} \cdot \Delta S}{(1+\beta)S}\right)$ for the compressed mode.

$$N_C \approx \frac{PG}{\alpha(1+\beta)(E_b/I_o)_{target}} + 1 - \frac{FN_{th}W}{\alpha(1+\beta)S} - \frac{M\Delta S}{(1+\beta)S} \tag{3-2}$$

The more users in the compressed mode the more capacity will be degraded. So a capacity-based compressed mode is proposed to reduce the impact.

According to equation 3-2, Figure 3-1 shows the capacity with varied interference. The capacity with the compressed mode is obviously lower than the capacity without the compressed mode. Initially the scenario of many users with the compressed mode is located at point a. The interference exceeds the maximum tolerated value and the capacity is only C_{CM}. In order to enhance the capacity, the number of users with the compressed mode is reduced and the relation curve tends to the non-compressed mode curve gradually (the operating point towards b). The interference is then reduced and the capacity can increase toward C_{NCM} (as point c). Either the number of users in the cell is reduced (as point d), and the interference is low enough to restart the suspended compressed mode. With the suspended compressed mode users operating the compressed mode again, the operating point tends toward point e. Last, when the number of users increases, the operating point backs to point a. This plot announces that arranging the compressed mode operation can effectively improve the capacity.

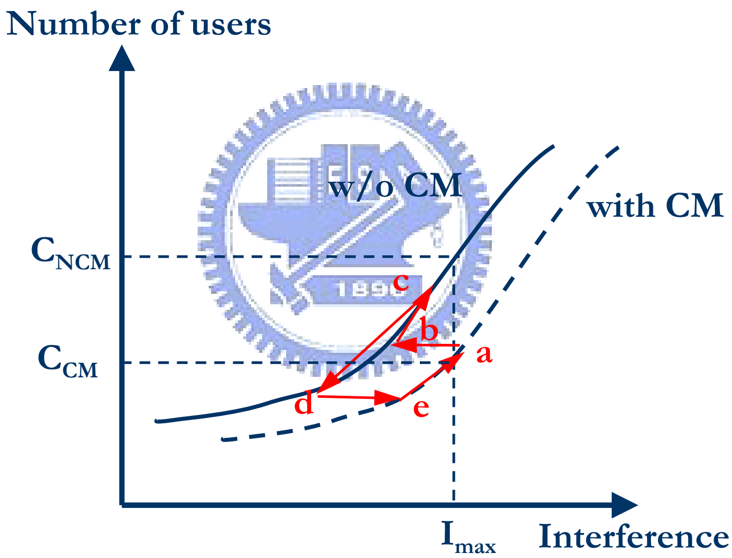


Figure3- 1. The capacity with different interference

When concerning about the downlink capacity, the downlink capacity is decided from the maximum power budget as equation (3-3):

$$P_{tot} = P_{OH} + \sum_{N}\left[\alpha \cdot P_i(\left(\frac{E_b}{I_o}\right)_{target}) \cdot M_i(CM)\right] \leq P_{\max} \tag{3-3}$$

where $\boldsymbol{P_{tot}}$ is the total base station transmitted power, $\boldsymbol{P_{OH}}$ is the overhead power, $\boldsymbol{N}$ is the number of users in the cell, $\boldsymbol{\alpha}$ is the voice activity, $\boldsymbol{P_i}$ is the base station transmitted power for single user, $\boldsymbol{M_i}$ is the multiplier of compressed mode power increase, and $\boldsymbol{P_{max}}$ is the maximum power budget. The $\boldsymbol{P_i}$ is decided by the target bit energy to interference ratio

$\left(\frac{E_b}{I_o}\right)_{target}$ and the M_i is decided by whether the compressed is operated or not. M_i is equal to 1 when the compressed mode is not operated. When operating the compressed mode M_i is equal to the multiple of power increase. The total transmitted power is required to be less than the maximum power budget. It can be observed that the more power increases in the compressed mode the less capacity could be supported. Whether the uplink or the downlink, the capacity is decreased when the number of the compressed mode users is increasing. Consequently, a capacity-based compressed mode is required for both uplink and downlink transmission.

3.3 The Compressed Mode Format

The compressed mode format is composed of gap generation method, triggering criteria and gap pattern. In the simulation, these factors are considered as below:

a. Gap generation method

As discuss in chapter 2: Puncturing cannot be used in uplink, and can only generate small gaps. Scheduling cannot apply to real time service. Only reducing the spreading factor by two is chosen into simulation among the three methods and can be suitable for all 3G services.

b. Compressed mode triggered criteria

The pilot Ec/Io is the typical threshold for the UMTS soft handover. However the pilot Ec/Io is not suitable for the compressed mode triggering due to following two reasons. The first reason is the border cell effect which means the Ec/Io decays smoothly when the mobile away from the base station. In border cells, the pilot signal and the interference are under the same fading condition, thus the pilot Ec/Io will hold the value until hitting the noise limit, where the total interference is dominated by the background noise. Figure 3-2 shows the different Ec/Io degrade scenario between the center cell and the border cell. It can obviously see the curve keeps consistence in the border cell and the pilot Ec/Io can not reflect the signal strength. The border cell effect is also illustrated in [31] and the curve also follows the same trend. The second reason is that the pilot Ec/Io is influenced by the different loadings. Figure 3-3 shows the pilot Ec/Io varies with different loadings. In the center cell, the UMTS uses relative threshold to trigger handover. However in the border cell, the mobile might measure only one pilot signal. As a result, the algorithm hardly finds a proper absolute threshold adapt to all the different loading. In this case, the pilot Ec/Io is not suitable for being the compressed mode triggering and handover decision.

To ensure in-time measurement and avoid unnecessary compressed mode triggering, the event triggering by pilot RSCP is chosen. For a baseline performance of measurement, the simulation starts the compressed mode when the pilot RSCP is smaller than -95 dBW and stop the compressed mode when the pilot RSCP is larger than -90 dBW. If the mobile keeps on going out, it will hand-down to GSM cells. The handover triggers when the RSCP is smaller than -118 dBW for 500 ms. The trigger timer is used for avoiding the ping-pong effect. The above parameters are listed in Table 3-1. The threshold settings are according to the distance from the base stations with zero fading.

Table3- 1. Compressed mode triggering threshold

Threshold	Value	Distance
Compressed mode start threshold for pilot RSCP	-104 dBW	0.5*radius
Compressed mode stop threshold for pilot RSCP	-108 dBW	0.6*radius
Handover triggering threshold for pilot RSCP	-118 dBW	radius
Time to trigger handover	500 ms	

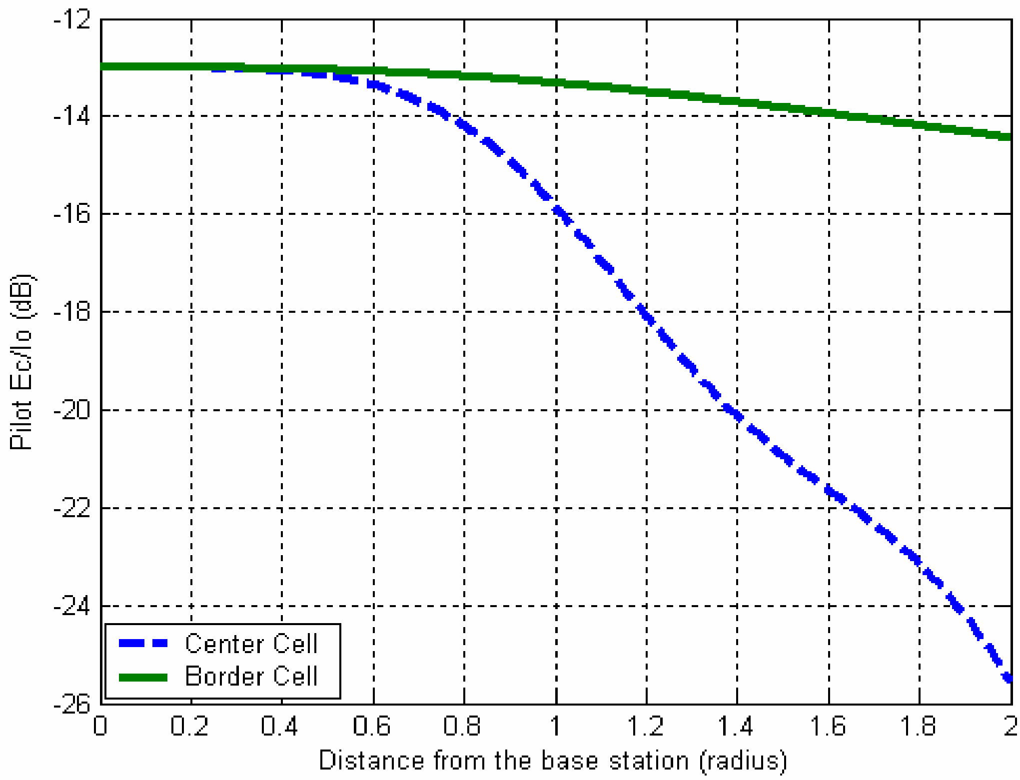


Figure3- 2. The pilot Ec/Io decay curve in center and border cells

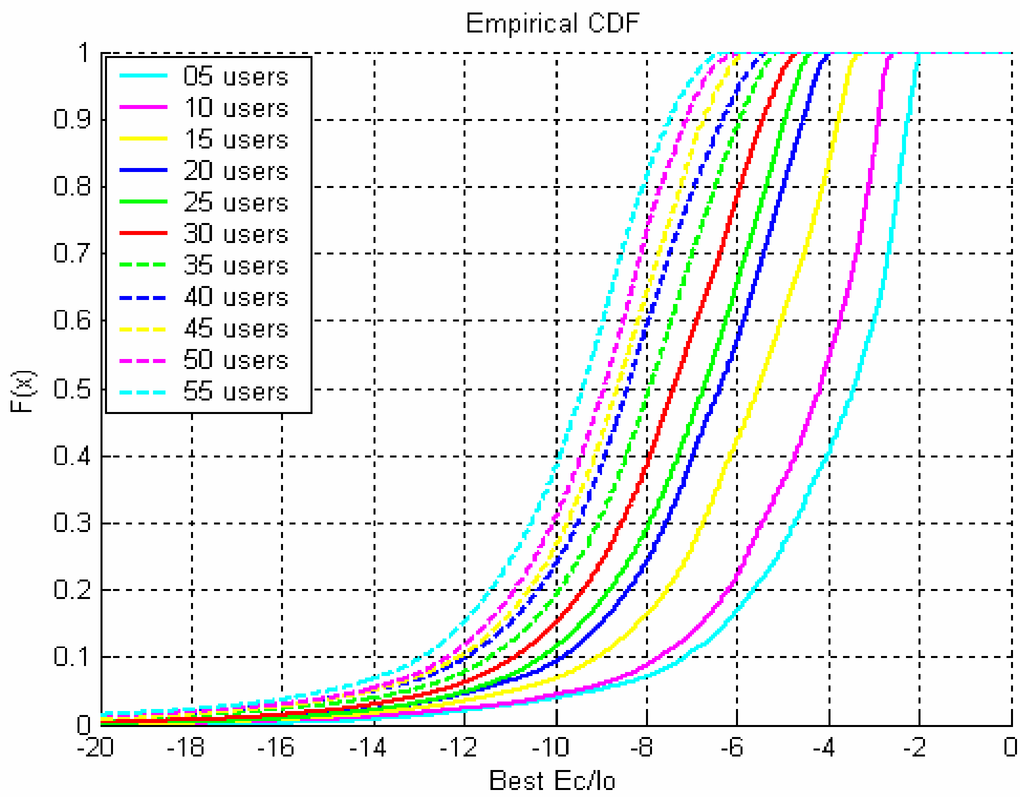


Figure3- 3. The pilot Ec/Io distribution function with different londing

c. Gap pattern to measure GSM carrier

In GSM, only Frequency Correlation Channel (FCCH), Synchronization Channel (SCH) and Broadcast Channel (BCH) are transmitted at all time. To be useful, the measurement of GSM carriers and Base Station Identity codes (BSICs) [14] should only on FCCH and SCH. However, the gap patterns in UMTS specification [14] are not guaranteed to match with the GSM timing structure. Ideally, the measurement gap should be 9.2ms for every 46ms, but the formula does not match UMTS format. In the simulation, the gap time of 14 slots for every 5 frame (9.3ms for every 50ms) is chosen to approach the GSM control frame structure. The GSM channel scenario is depicted in Figure 3-4(a) and the corresponding gap pattern is depicted in Figure 3-4(b).

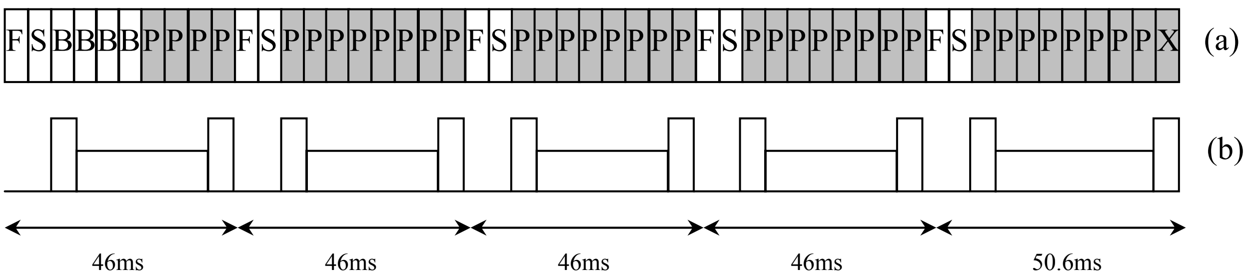


Figure3- 4. (a) GSM control channel (b) Compressed Mode gap pattern

3.4 The Algorithm of the Capacity-Based Compressed Mode

When a mobile needs to measure other systems such as GSM, the mobile can measure only at few measurable time slots. As a result, all the users in border cells will execute the compressed mode simultaneously to match the measurable time slots as depicted in Figure 3-5(a). The increasing aggregate power could cause a serious impact on the capacity. To resolve the power problem, two methods are suggested and are depicted in Figure 3-5(b) and 3-5(c). The first one is to separate the position of transmission gaps and spreads out the aggregate increasing power. The second one is to schedule the order of the execution of the compressed mode. However, the first one moves the transmission gap to the adjacent time slot but there is no guarantee that the new measurement interval can match with the actual transmission slots of other systems. As a result, the second method is chosen for the proposed algorithm.

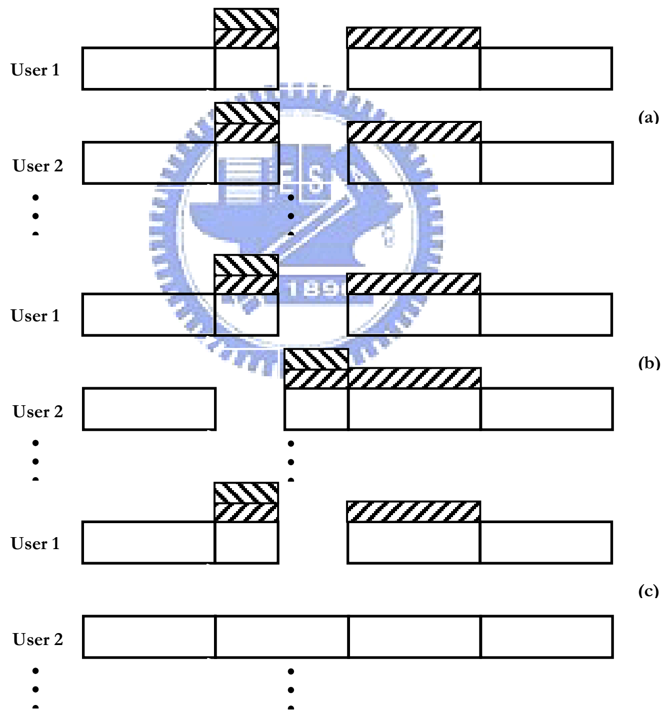


Figure3- 5. The compressed mode scheme (a) Normal compressed mode at simultaneous time (b) Separate the position of transmission gap (c) Schedule to suspend the compressed mode

First, the relationship of pilot RSCP versus the distance from the base station is depicted in Figure 3-6. When the mobile is close to the base station, the curve shows the exponential increase of the RSCP. When the mobile is away from the base station, the curve tends to stay linear. By using the linear relationship, the critical pilot RSCP ratio, R_{RSCP}, to represent the distance that needs performing the compressed mode is calculated in equation (3-4).

$$R_{RSCP} = \frac{T_{stop} - RSCP}{T_{stop} - T_{ho}} \tag{3-4}$$

where T_{stop} is the threshold to stop the compressed mode, and T_{ho} is the threshold to trigger border-cell handover. According to the previous compressed mode format, the compressed mode operates only in between T_{stop} and T_{ho} (R_{RSCP} is ranged in between 0 to 1). The compressed mode stops when R_{RSCP} equals to 0. The hand-down to GSM occurs when R_{RSCP} equals to 1. The relationship between R_{RSCP} and RSCP is shown in Figure 3-7. According to R_{RSCP}, it can estimate the proportion of the effective distance for the compressed mode operating.

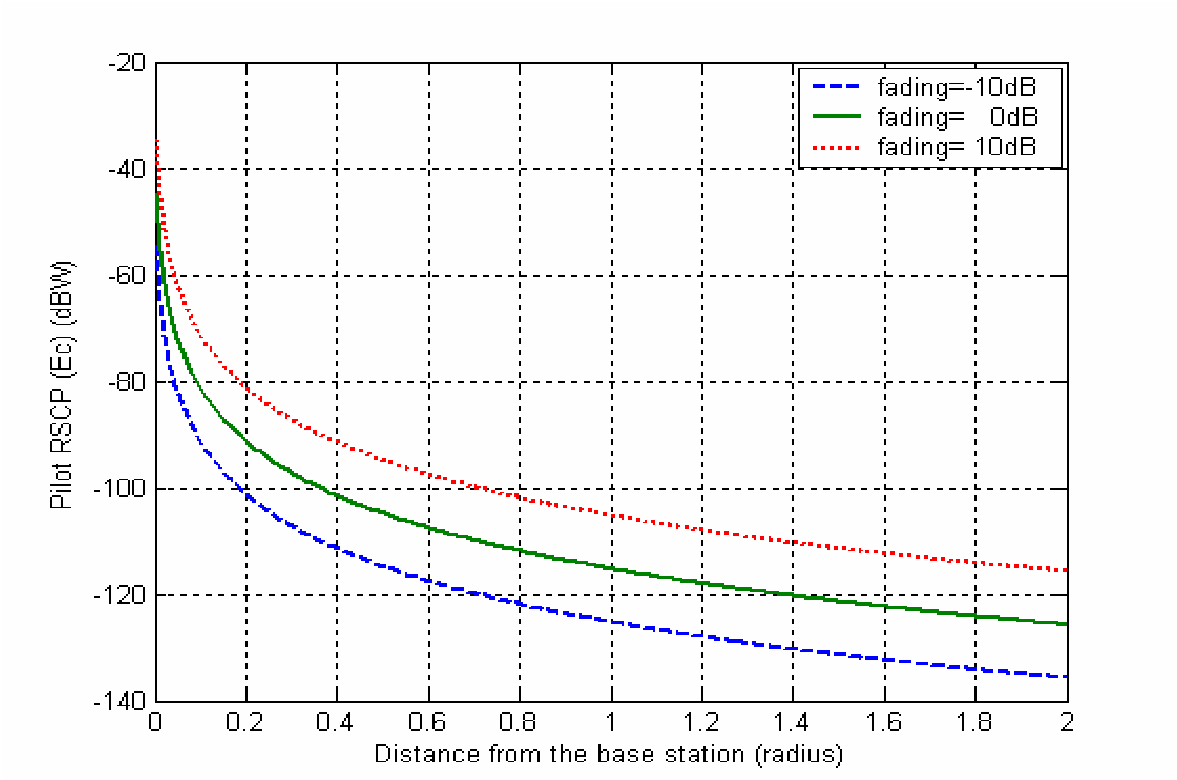


Figure3- 6. The relationship of Pilot RSCP and distance

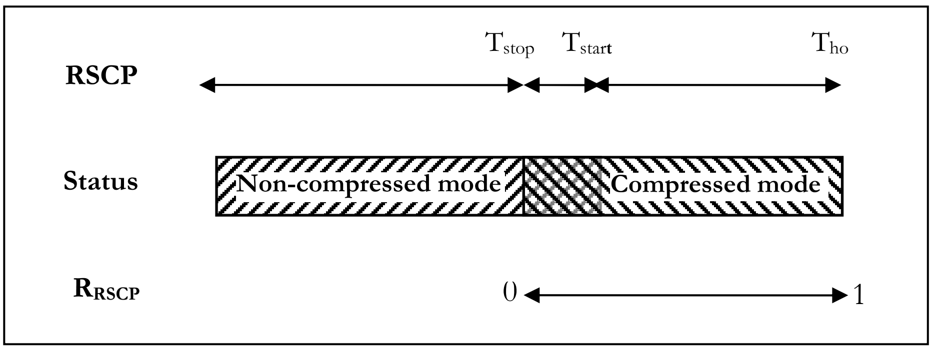


Figure3- 7. The relationship of R_{RSCP} and distance

The proposed capacity-based compressed mode limits the power level in the compressed frame by suspending low priority users from operating the compressed mode. Thus it can ensure the capacity while maintaining the proper hand-down priority. The following steps of this algorithm are described as follows:

(1) Every frame, the suspend factor $\boldsymbol{F_i(n)}$ base on the pilot RSCP ratio $\boldsymbol{R_{RSCP}}$, the aggregate measured GSM samples $\boldsymbol{N_{meas}(n)}$, and the record of last time compressed mode $\boldsymbol{R_{suspend}}$ is calculated in equation (3-5).

$$Suspend \quad factor \; : \; F_i(n) = \frac{N_{meas,i}(n-1)}{R_{RSCP,i}(n)^k} \times R_{suspend,i}(n) \tag{3-5}$$

$$R_{suspend,i}(n) = \begin{cases} 1, & last \;\; compressed \;\; mode \;\; has \;\; been \;\; suspended \\ 2, & otherwise \end{cases} \tag{3-6}$$

As shown, the algorithm prefers to suspend the users close to the base station (small $\boldsymbol{R_{RSCP}}$). The users keep larger sampled GSM carriers, $\boldsymbol{N_{meas}}$, will easily be suspended. The schedule algorithm wants to make each user have the same $\frac{N_{meas}}{R_{RSCP}{}^k}$; it can balance the aggregate measured samples with the effective distance for the compressed mode operating. Besides, there is a tunable factor k to modify the schedule algorithm. The factor k modifies the dominant degree of the distance. Finally, the $\boldsymbol{R_{suspend}}$ equals to 1 when the last time compressed mode operation have been suspended, otherwise it equals to 2 as equation (3-6). This term halves the suspend priority for users who just has been suspended and is designed to reduce the chance of the continuous suspension of the same user. The continuous suspension may delay the measurement efficiency and it might be delayed to handle the emergency handover.

(2) The second step is to observe the base station transmitted power and set a suspend threshold of transmitted power $\boldsymbol{P_{thr}}$, which is smaller than the maximum transmitted power budget. If the estimated base station transmitted power, $\boldsymbol{P_{BSest}}$, doesn't exceed the threshold $\boldsymbol{P_{thr}}$ and then the system can operate the compressed mode without any suspension to guarantee the measurement efficiency. If the $\boldsymbol{P_{BSest}}$ exceeds $\boldsymbol{P_{thr}}$, then the system suspends the compressed mode according to the priority order based on $\boldsymbol{F_i(t)}$ until the transmitted power below $\boldsymbol{P_{thr}}$ or there is no other compressed mode available.

The flow chart of the capacity-based compressed mode is depicted as Figure 3-8. The base station collects the information of the compressed mode and computes the suspend factor for all users in the compressed mode on every frame and schedule the users to suspend their compressed mode operation. Then, the base station examines the transmitted power and decides whether to suspend the compressed mode. As a result, the proposed algorithm reduces by suspending better RF users to maintain the system capacity.

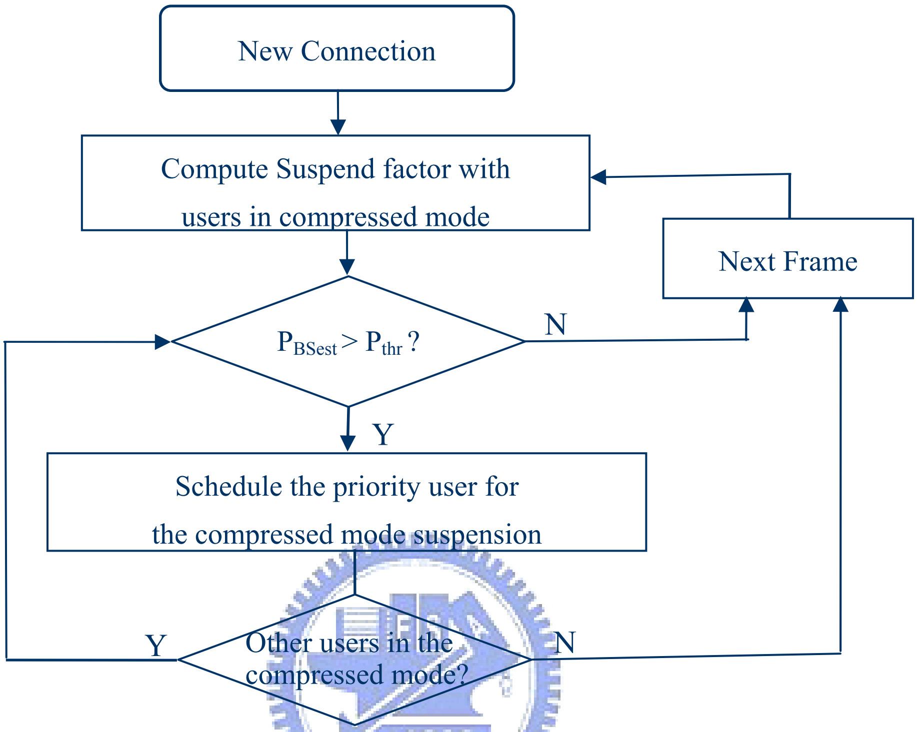


Figure3- 8. The flow chart of capacity-based compressed mode

In this study, the proposed control algorithm will consider only the downlink capacity. The reasons are as follows:

1. Only the base station can acquire the information of all users' compressed mode profiles. The base station executes the capacity-based compressed mode and sends messages to all mobiles in the cell to control the compressed mode.
2. In the asynchronous data transmissions, the required throughput of the downlink is much higher than the uplink. As a result, the capacity limit is on the downlink in the multimedia service. The downlink transmission is required to propose a control algorithm to take care its performance.
3. In most cases, the compressed mode is used in the downlink. Since the base station can stops the transmission and let the mobiles have the free time to measure the signal strength of other systems. If both the downlink and uplink compressed mode is performed, the addition time-aligned with time offset is needed.

Chapter 4

Simulation Platform

The proposed capacity-based compressed mode has been introduced. It needs a simulation platform to verify the performance. This chapter introduces the simulation platform and the platform is constructed of 19 UMTS cells. The cell radius is calculated by the uplink link budget and the propagation model. The details of the environment establishment are addressed here. Moreover the capabilities of the platform are also presented.

4.1 Simulation Environment and the Mobility Model

This simulation model constructs UMTS wireless communication system scenarios. There are 19 UMTS cells each with 3 sectors and the UMTS is surrounded by GSM Sea as depicted in Figure 4-1. Each cell has three antennas and the angles of the antennas are 0°, 120°, and 240°. In each UMTS sector, the base station will execute the soft handover and downlink power control to support the mobility. When the moving mobile is away from the serving sector, the signal strength is weakened and the connection quality is degraded. The soft handover connects to other sector to guarantee the seamless connection, and the downlink power control ensures the quality of the received signal to interference ratio (SIR).

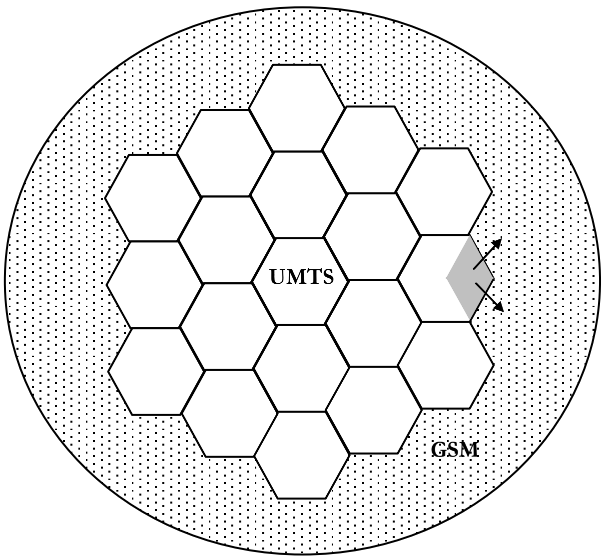


Figure4- 1. The Simulation Platform

This platform can simulate with arbitrary numbers of mobile within the supported capacity in each sector. It only observes the mobiles in the border sector. Initially, all of the mobiles are random placed in the border sector with uniform distribution. Each mobile is assumed keeping the same moving speed during the simulation period. The moving speeds are also uniform distribution range from 50 to 100 km/hr. After each sample period, each mobile changes the moving direction in a Gaussian distribution with mean equals to zero and the standard deviation equation to 1.6°. The slight changing direction affects the reality human behavior. The simulated trajatory is depicted in Figure 4-2.

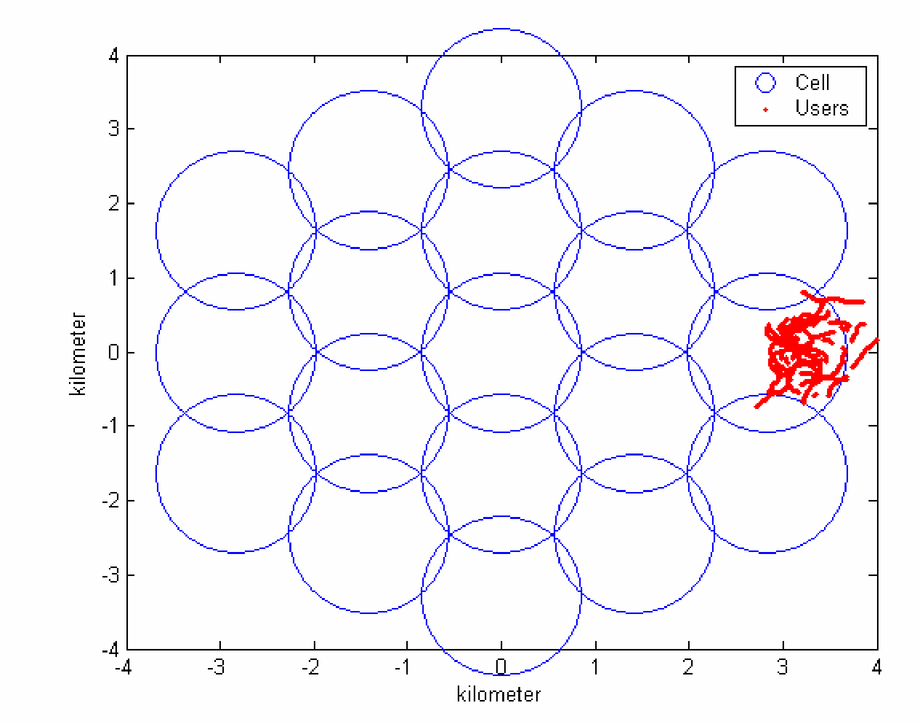


Figure4- 2. The mobility scenario

4.2 Uplink Link Budget and the Propagation Model

The second thing to establish the simulation platform is the cell size. It can estimate the coverage by the uplink link budget and the propagation environment. Within the coverage, all mobiles are guaranteed to transmit sufficient quality of the signal through the path loss. With the uplink link budget in Table 4-1, the maximum path loss can be calculated by transmitter gain, receiver gain, receiver sensitivity, soft handover gain, fading margin, and in-building penetration loss. The maximum tolerant path loss is then computed as 136.97dB.

Table4- 1. Uplink Link Budget

Uplink Link Budge			
Item	Symbol	Value	Units
Total TX power available	P_{TX}	21	dBm
TX antenna gain	G_{tx}	2	dB
Body loss	L_t	2	dB
TX EIRP per traffic channel	$P_{EIRP}= P_{TX} + G_{tx} - L_t$	21	dBm
RX antenna gain	G_{rx}	18	dB
RX cable and connector losses	L_{cable}	4	dB
Receiver noise figure	NF	3	dB
Temperature, kelvin	T	290	K
Boltzmann's constant	k_b	1.38×10^{-23}	(J/ k)
Thermal noise density	$N_{density}=10\log(k_b *T)$	-174	dBm/Hz
Noise rise due to interference	N_{inte}	4.56	dB
Total effect of noise	$N_{eff}= NF + N_{density}$	-171	dBm/Hz
Bit rate	R_b	12.2	kbps
Information rate	$R_{dB}=10\log(R_b)$	40.86	dBHz
Effective required E_b/N_o	E_bN_o	4.57	dB
RX sensitivity	$S_{RX}= N_{eff} + R_{dB} + E_bN_o$	-125.57	dBm
Soft handoff gain	G_{SHO}	4.5	dB
Fast fading margin	M_{ffad}	0.5	dB
Log normal fade margin	M_{log}	11.6	dB
In-building penetration loss	L_{pene}	16	dB
Maximum path loss	$PL_{max}= P_{EIRP} + G_{rx} - L_{cable} - S_{RX} + G_{SHO} - M_{ffad} - M_{log} - L_{pene}$	136.97	dB

Table4- 2. The parameter of the propagation model

The Propagation Model			
Item	Symbol	Value	Units
Path loss model		COST 231	
Frequency	f_c	2000	MHz
Base station antenna height	h_{bs}	30	m
Mobile station antenna height	h_{ms}	1.5	m

The longest distance that supports sufficient connection quality is derived from the known maximum path loss and propagation model. The propagation model is chosen as COST-231 Model and the parameters are listed in Table 4-2. The propagation geography model is depicted in Figure 4-3. The frequency band of UMTS communication system is 2000 MHz. The base station antenna height is assumed as 30 m and the mobile station antenna height is assumed as 1.5 m. These parameters are put into the COST 231-Model [32] as equation 4-1 & 4-2 to calculate the longest transmission distance in equation 4-3. The longest transmission distance is calculated as 0.95 km.

$$L(urban)(db) = 46.3 + 33.9\log f_c - 13.82\log h_{bs} - a(h_{ms}) + (44.9 - 6.55\log h_{bs})\log d \quad (4\text{-}1)$$

$$a(h_{ms}) = (1.1\log f_c - 0.7)h_{ms} - (1.56\log f_c - 0.8)dB \quad (4\text{-}2)$$

$$d = 10^{[(L - 46.3 - 33.9\log f_c + 13.82\log h_{bs} + a(h_{ms}))/(44.9 - 6.55\log h_{bs})]} \quad (4\text{-}3)$$

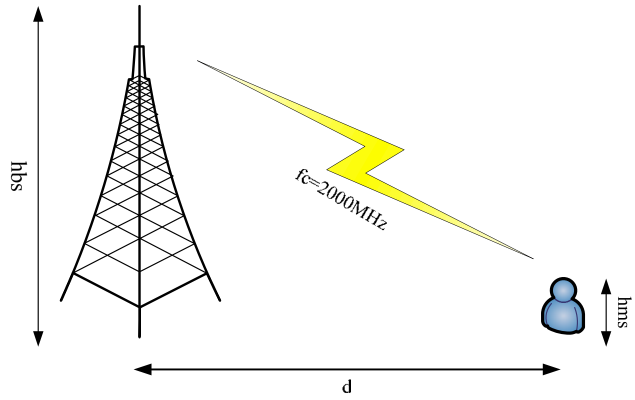


Figure4- 3. Propagation geography model

The cell geometric profile of the base station is a hexagon as depicted in Figure 4-4. The r1 is the unit distance in horizontal axis and r2 is the unit distance in vertical axis. The coverage is computed in equation 4-4. The longest path from the center point, r1, is equal to the longest transmission distance, d. The average coverage is equal to 0.88 km. The two-tier topology is then constructed with the same 19 hexagon cells.

$$r_1 = d \;\; ; \;\; r_2 = \sqrt{3}/2 \times r_1 \;\; ; \;\; r = \left((r_1 + r_2)/2 \right) \quad (4\text{-}4)$$

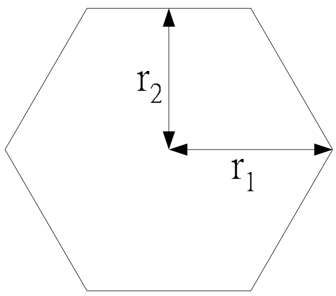


Figure4- 4. Base station cell geometric profile

4.3 The Platform Capability

The UMTS cells topology and the mobility model have been established. Based on the fixed converge, any arbitrary number of users is giving in the border sector. Each user transmits the same 12.2 kbps circuit switched services with the voice activity equal to 0.48. The maximum power budget is equal to 20 W including 4W overhead power. The simulation parameters are listed in Table 4-3.

Table4- 3. The simulation parameter

Simulation Parameter	
Radius	0.95Km
Mobile Speed	50-100 Km/hr
Bandwidth	3.84 Mcps
Bit Rate	12.2 kbps
Required Eb/No	4.57 dB
Voice Activity	0.48
Maximum Transmit Power	20 W
Transmit Overhead Power	4W (3W for pilot)

In this simulation platform, the number of users is modified to see the performance and then the capacity can be evaluated. The base station transmitted power, the pilot signal strength, and the compressed mode behaviors are also observed. Moreover this simulation platform could be easily modified to adapt for other wireless communication systems.

Chapter 5
Experimental Results

In the previous chapters, the capacity-based compressed mode algorithm has been proposed and the simulation platform has been introduced. In this chapter, the performance of normal transmission, regular compressed mode, and capacity-based compressed mode will be evaluated.

5.1 The Performance of the Capacity-Based Compressed Mode

Considering all users at all time, Figure 5-1 shows the average transmission power at different loadings. As expected, the regular compressed mode needs to transmit more power than the normal transmission. With the suspension control, the transmission power of the capacity-based compressed mode is in between the above two situation. By processing the capacity-based compressed mode, it can save the power consumption.

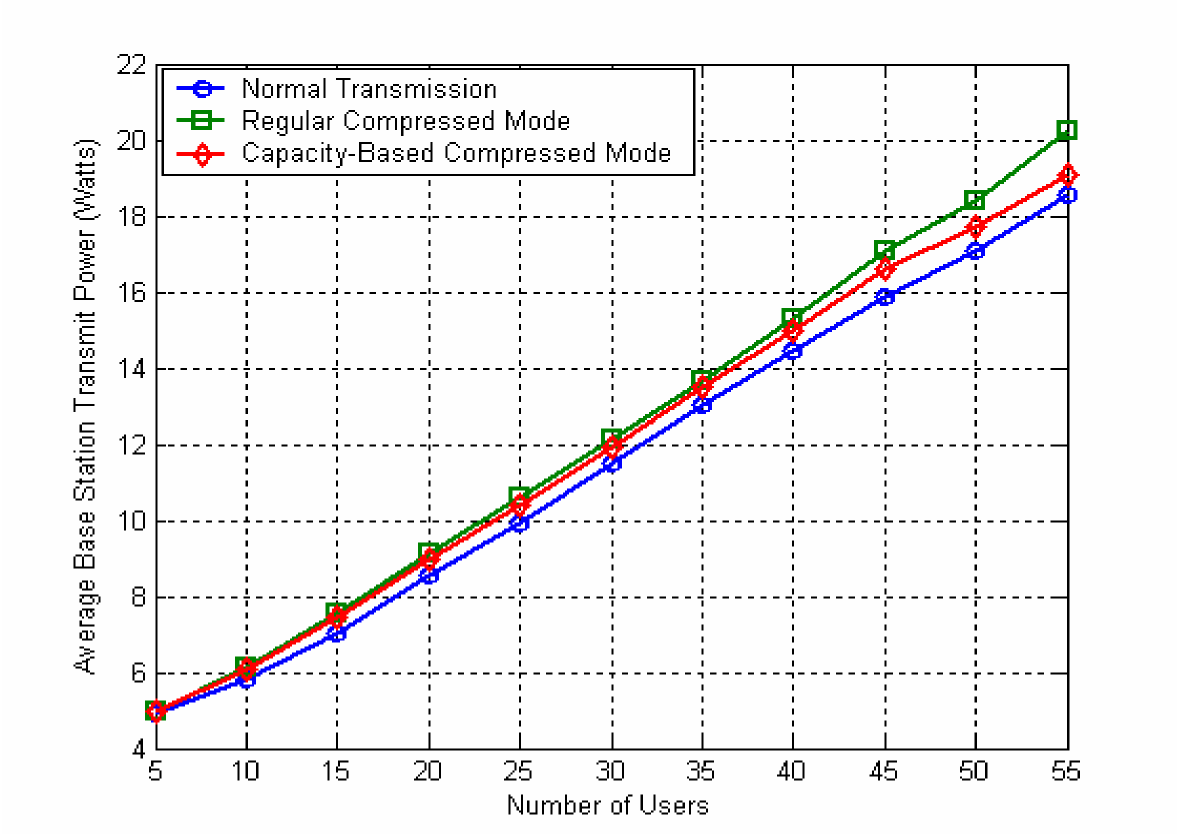


Figure5- 1. Average base station transmit power

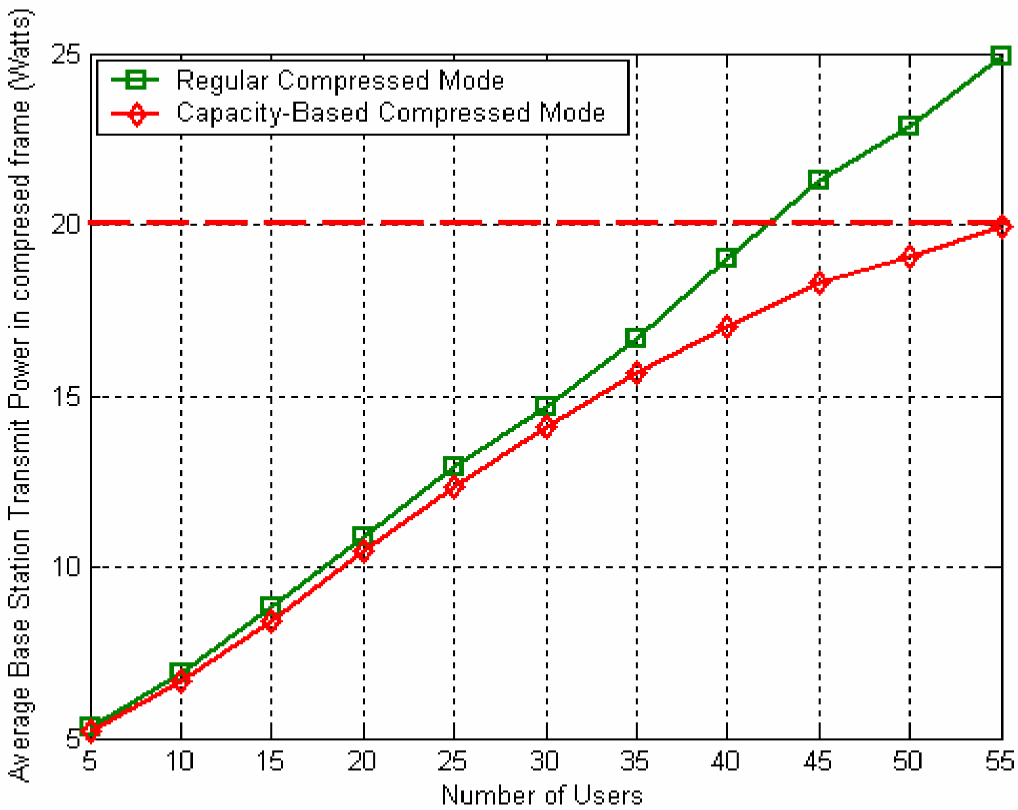


Figure5- 2. Average base station transmit power in the compressed frame

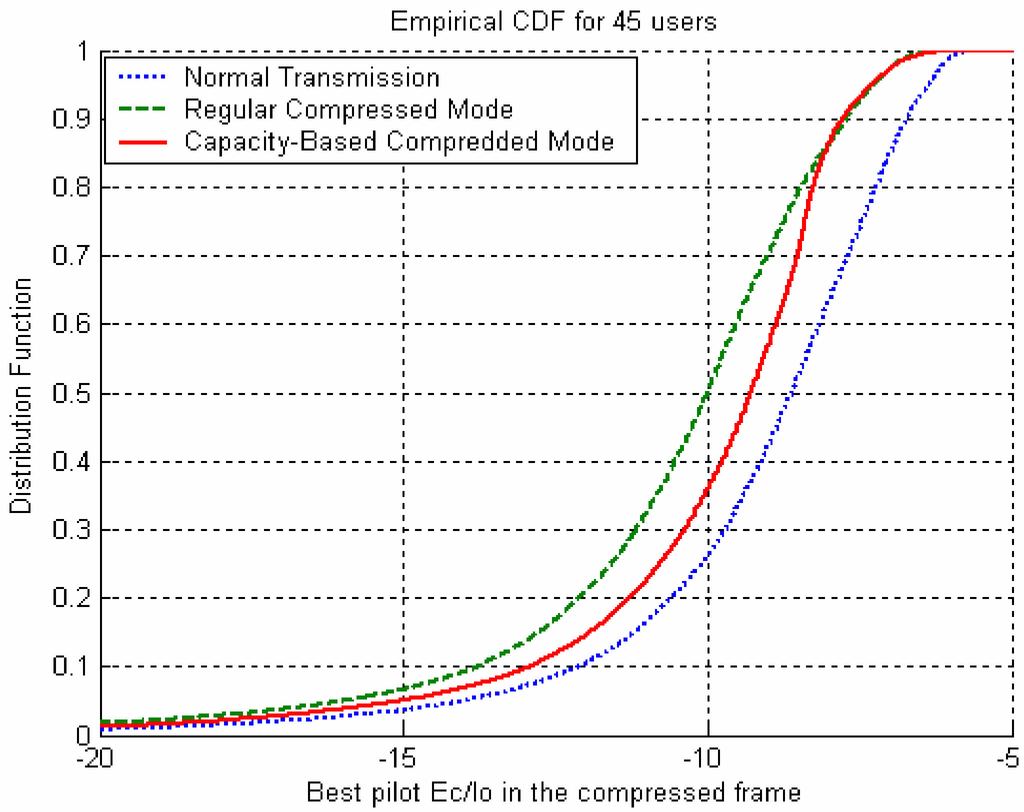


Figure5- 3. The best pilot Ec/Io distribution function for 45 users

As the loading increases, the proposed capacity-based compressed mode will increase the chances of the suspension. As shown, the base station power will close to the ‘Normal Transmission’ where there is no user in operating the compressed mode.

Since the compressed mode only operates one measurement per 5 frames. To understand the instant impact from the compressed frame, Figure 5-2 focuses only on the average power of the compressed frames. As shown, the power of regular compressed mode exceeds the tolerable maximum power budget (20 watts) and reaches performance break point when the number of user is larger than 42. Using the proposed algorithm, the capacity can be improved to 55. Moreover, the power saving is ranged from 2% to 20% while the loading is from 5 to 55. It is worth emphasizing here that the proposed capacity-based compressed mode is designed mainly to suspend better RF users when the traffic loading is high.

It is intuitive that the Ec/Io will be degraded when the total power is increased. From Figure 5-2, the regular compressed mode can support only 45 users at most. To compare the Ec/Io performance, the total number of users is set at 45. As shown in Figure 5-3, the best pilot Ec/Io of the regular compressed mode is less than the normal transmission by 1dB to 2dB. The best pilot Ec/Io of the proposed compressed mode is between the former two. Especially, the proposed algorithm improves the best pilot Ec/Io in the worse RF situation. This is because the suspensions are operated when the interference is higher. Consequently, the proposed algorithm enhances the signal strength when the signal strength is weak.

Figure 5-4 shows the suspension ratio with different loading. As shown, the suspension ratio increases when the number of users increases. Clearly, it trades off the transmitted power with suspension ratio. Obviously the concern is whether the high suspension ratio will degrade the handover efficiency.

Figure 5-5 shows the number of GSM Received Signal Strength Indicator (RSSI) samples before handing down to GSM systems. The ratio is less than the suspension ratio in Figure 5-4. This is because the proposed algorithm suspends mostly the relative good RF users from operating the compressed mode. For those users who has poor RF will be allowed to operate the compressed mode.

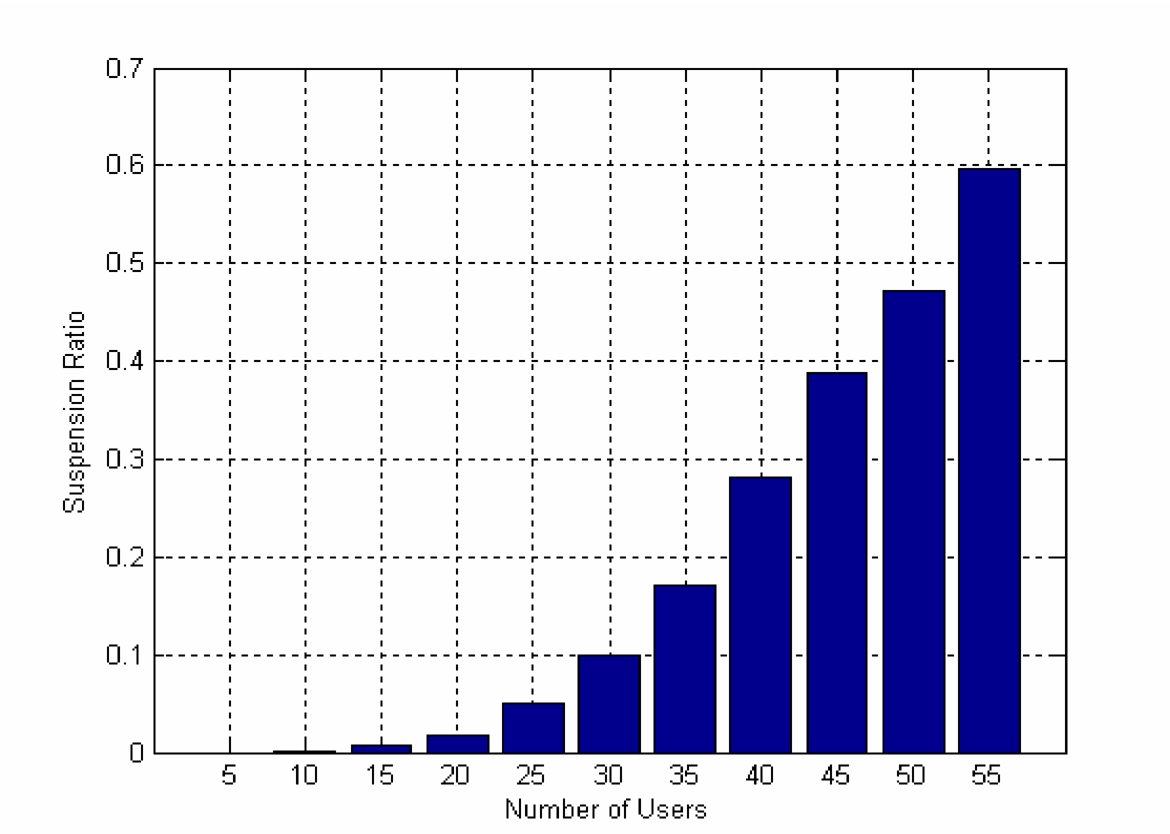


Figure5- 4. The suspension ratio with loading

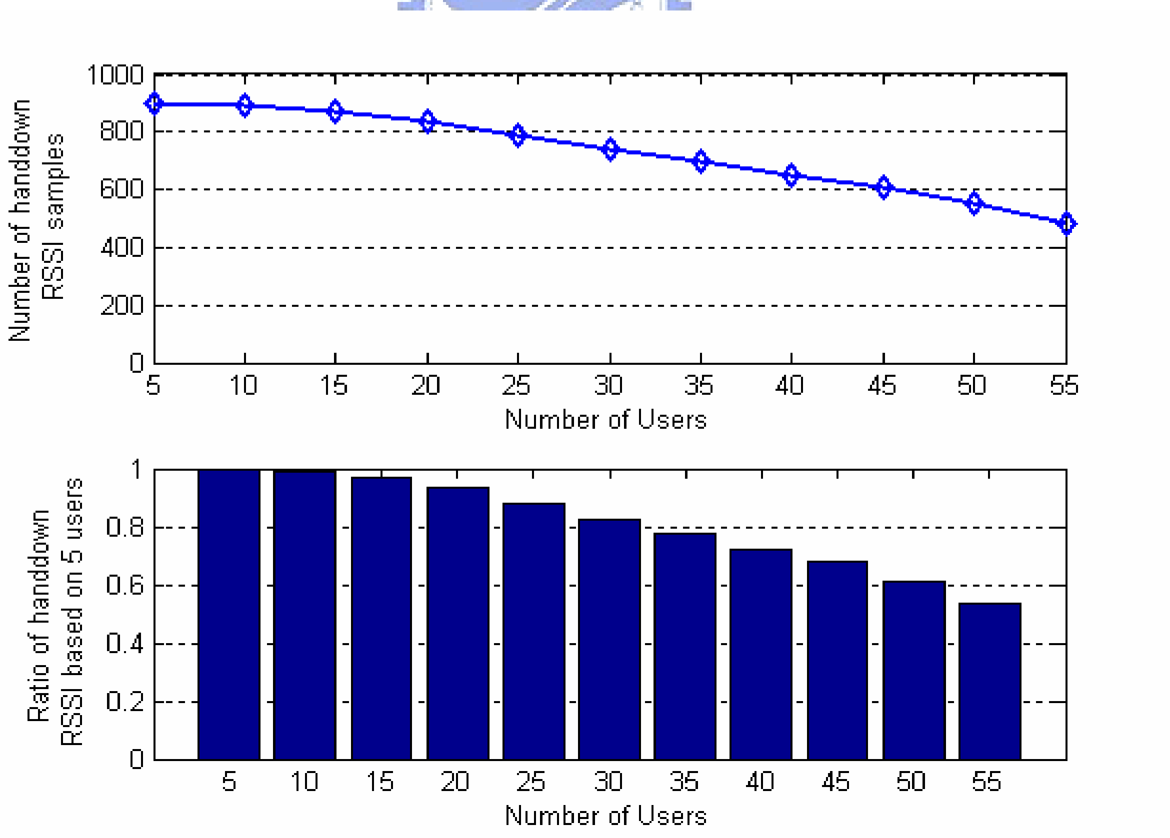


Figure5- 5. The number of hand-down RSSI samples with loading (k = 2)

Assuming the maximum 32 carriers in GSM neighbor list and the first 8 carriers are into BSIC verification and re-verification processes [21]. For each process, it needs at lease 3 samples. So, the expected minimum number of GSM carrier samples is equal to (32+8+8)*3=144. From Figure 5-5, the number of RSSI samples is enough for handover (larger than 144) even at high traffic loading. This is because the transmission gaps always match the measurable time and the system doesn't execute the useless compressed mode.

5.2 The Performance of the Factor k

At last, the tunable factor k of suspend algorithm in equation (3-5) is evaluated in Figure 5-6. The average number of the handed-down GSM RSSI samples and the variance are illustrated in the figure. The factor k is the weighting factor about the influence of RSCP ratio.

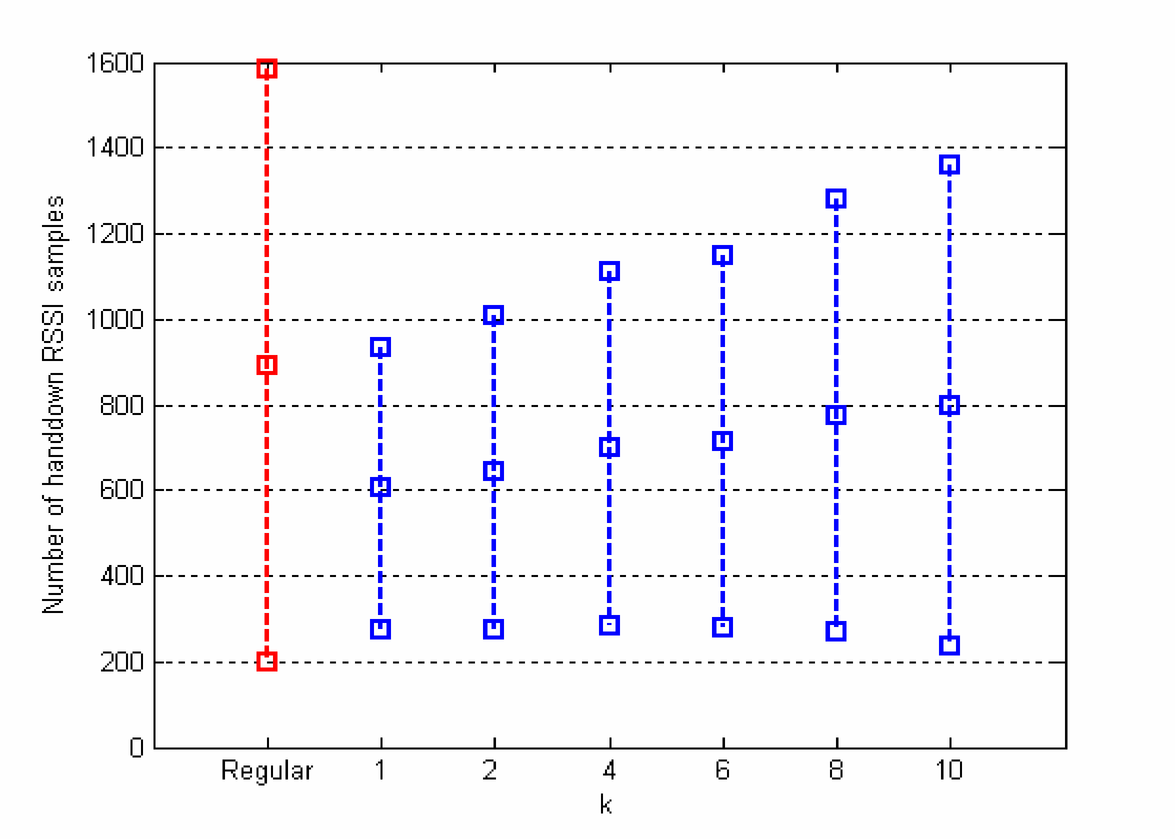


Figure5- 6. The number of hand-down RSSI samples with different k (User = 45)

The increasing k will force the suspension goes to users who are close to the base station easily and make it harder to suspend users who are at the hand-down region. As shown in Figure 5-6, the number of RSSI measurement samples before GSM hand-down will increase as the k increases. In other respect, high k makes the distance (or the associate RF) have bigger impacts on the selection of the suspend users. As a result, in a mobility environment, the variance of the RSSI measurement will also increase for higher k. The factor k trades off between measurement efficiency and fairness.

As shown in Figure 5-7, the relationship of the suspend probability and the strength of RSCP is illustrated. The suspend probability between the stop threshold (-104 dBW) and the start threshold (-108 dBW) is relative small. Besides this region, the larger RSCP (i.e. the RF condition is better) has higher opportunity to be suspended than the lower RSCP. With the modified factor k increasing, the RF condition is more dominated and this phenomenon is becoming obvious. It can explain in Figure 18 why the number of the hand-down samples is higher when the k is larger. The factor k can easily modify the importance of the RF influence.

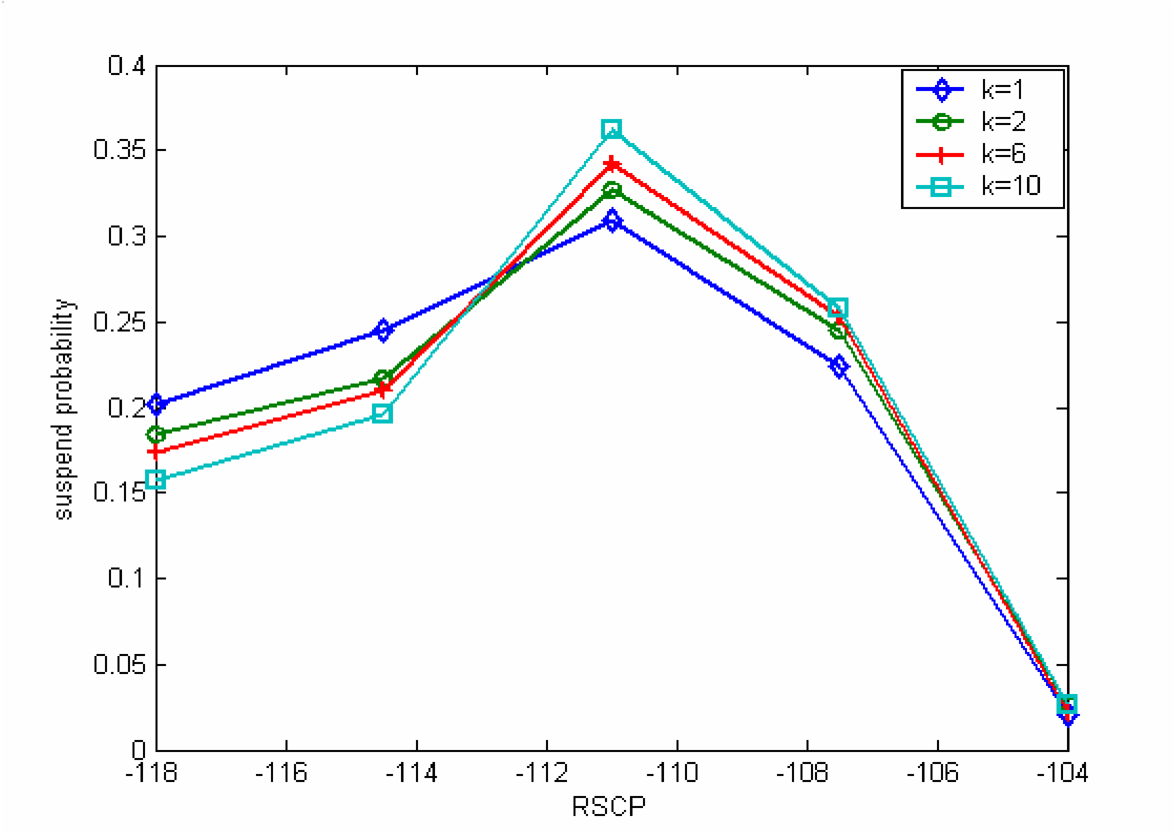


Figure5- 7. The relationship of the suspend probability and RSCP with different k (User = 45)

Chapter 6
Conclusions

6.1 Contributions

In this thesis, the capacity-based compressed mode is proposed to resolve the excessive power increase during the compressed mode. To differentiate the priority users based on RF condition, number of RSSI measurement, and continuous suspension avoidance, the proposed algorithm can effectively reserve the capacity by reducing the number of simultaneous compressed mode measurements while keeping the more-than-enough measurements before a user handing-down to GSM system. Furthermore, with different choices of k factors, users with different priority in terms of the necessity of border-cell hand-down can be easily separated. The proposed capacity-based compressed mode control can apply to a general design concept for trading off the capacity and handover performance if users' priority can be identified. The contributions of this thesis are summarized as:

1. Study and report on the detail operations of the compressed mode to execute inter-system handover.
2. The proposed capacity-based compressed mode improves the performance impacts on the capacity caused by the compressed mode.
3. The simulation platform is established for both intra-system handover and inter-system handover in UMTS system.
4. The simulation platform verifies the proposed control algorithm truly works and it can be applied to handover to other systems.

6.2 Future Works

In this thesis, the simulation only considers about the same 12.2 kbps circuited switched services. In the future, the different QoS services are occurred and it will challenge this algorithm. Since the proposed capacity-based compressed mode only considers the measurement issue. For making handover decisions, no matter what services needs the enough carriers' information of other system. Thus, the algorithm is still work but it will add further criteria in the handover stages. Like the ideas in [33], the handover ordering is decided by the signal strength and the service profile. The next step, this algorithm can consider about the multimedia services and add new control criteria to inter-system handover.

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