# Guaranteeing Quality of Service in Mobile Radio Networks by Means of Link Adaptation Algorithms

Javier Gozalvez and Miguel López-Benítez
Signal Theory and Communications Division
University Miguel Hernández
Avda de la Universidad s/n, 03202 Elche, Spain
j.gozalvez@umh.es

Oscar Lázaro
Mobile Communications Group
Universidad Politécnica de Valencia (UPV)
Camino de Vera S/N, 46022 - Valencia, Spain

*Abstract*— **Link Adaptation has been proven to be an efficient technique to enhance the performance of mobile communication systems. The design of Link Adaptation has mainly concentrated on improving throughput performance. Although this design strategy can be considered appropriate for best effort services, multimedia services have important constraints in terms of transmission delay and error rates that can be tolerated without considerable degradation in the Quality of Service that the user might experience. In this context, this paper presents Link Adaptation algorithms that are able to control the experienced error rate and also to trade-off between throughput and error rate. The proposed algorithms are evaluated for real-time H.263 video transmissions in a packet-switched mobile radio network based on the GPRS standard.**

***Keywords-Link Adaptation, Quality of Service***

## I. INTRODUCTION

The main objective of Radio Resource Management (RRM) techniques is the efficient use of the scarce available radio resources. Such efficient use is essential for operators considering the high demand for traditional voice services, the introduction of new bandwidth-consuming data and multimedia services and the increasing importance of providing services with an appropriate Quality of Service (QoS). Link Adaptation (LA) is an adaptive RRM technique that has already proven its benefits in terms of improved speech quality, coverage and throughput [1]. Its potential and benefits are such that it is considered as a key technology for evolved 2G and 3G systems.

Link Adaptation seeks to efficiently use the scarce available radio resources by adaptively selecting a suitable transport mode (e.g. modulation and/or coding scheme) according to the experienced channel quality conditions. Thus, a key issue in the design of LA algorithms is the actual criteria used to select the optimum transport mode for the experienced radio link quality conditions. A commonly used criteria is to select the transport mode that maximises the throughput [2]. Although this criteria might be appropriate for best-effort services, its use has been questioned for multimedia services with real-time or tight transmission delay requirements and error performance constraints. As a result, several other criteria used to decide the optimum transport mode in the case of multimedia services have been proposed in the literature. Ref. [3] proposes a LA algorithm designed so that the transport mode selected maximises the video quality measured in terms of the Peak Signal to Noise Ratio. The work reported in [4] presents a LA algorithm designed to reduce transmission delays. Such algorithm bases its decision on the size of the packet to be transmitted as transmission delay depends on such size. On the other hand, the algorithm proposed in [5] also seeks to reduce transmission delays but by directly including such performance measure in the adaptation process. Since certain multimedia services need low packet error rates for not degrading the user perceived QoS, a different approach for the design of LA algorithms is considered in [6]. In particular, the LA algorithm evaluated in [6] is designed to achieve a target error rate for a music streaming service.

Although the algorithm proposed in [6] manages to reduce the packet error rate experienced in downlink transmissions compared to an LA algorithm whose criteria is to maximise the system throughput, the results obtained are far from the set error rate target. In this context, this paper proposes a simple LA algorithm to actually further reduce the transmission errors and guarantee that the specific error rate target required by certain multimedia services is reached. In particular, this work considers real-time H-263 video transmissions. While the proposed LA algorithms achieve the expected error rate target at the expense of system throughput, throughput-based LA algorithms increase system throughput at the expense of error rate performance. This paper also proposes a simple LA algorithm that allows to trade-off between throughput and error rate performance, giving the operator the ability to adjust the LA algorithm according to its needs and the expected QoS.

## II. EVALUATION ENVIRONMENT

### A. *General Packet Radio Services (GPRS)*

The work reported in this paper has been conducted for packet data transmissions in a GPRS-like system. Although the low transmission rates achieved by GPRS may raise some questions about its feasibility to transmit video, it is worth mentioning that the aim of this research is not to prove the feasibility of video transmissions over GPRS but to propose LA algorithms that improve and guarantee the QoS of video transmissions.

This study has focused on the RLC/MAC and physical layers of the GPRS standard. Prior to transmission, data

This work has been sponsored by Bancaja and the University Miguel Hernández.

TABLE I. GPRS CODING SCHEMES

| Scheme | Code rate | Payload | Data rate (kbits/s) |
|---|---|---|---|
| CS1 | 1/2 | 181 | 9.05 |
| CS2 | ≈2/3 | 268 | 13.4 |
| CS3 | ≈3/4 | 312 | 15.6 |
| CS4 | 1 | 428 | 21.4 |

packets are segmented across the different layers of the GPRS protocol stack, with the final logical unit being the RLC block. The resulting RLC data blocks are then coded and block-interleaved over four normal bursts in consecutive TDMA frames. The actual information transmitted in a RLC block depends on the particular Coding Scheme (CS) used to protect the information from radio transmission errors. Although GPRS considers a single modulation scheme, it defines four different CS; see Table 1. These four CS offer a trade-off between throughput and coding protection, paving the way for the application of dynamic LA to GPRS.

### *B. System Level Simulation Platform*

The performance evaluation of a cellular system is usually conducted at two different levels: system level and link level. While the former models a mobile radio network, the later models the radio link at the bit level. Since this study concentrates on system level aspects of the operation and performance of LA algorithms, this section briefly presents the system level simulation tool that has been employed. The necessary interfaces between the link and system level studies are presented in the next section.

This work has been conducted using a highly accurate event-driven simulator working at the burst level [7]. The simulator models a sectorised macrocellular network and concentrates on the downlink. Users are assigned channels in a first-come-first-served basis by means of a random allocation scheme. In this paper, only single slot transmissions have been considered. User mobility has been implemented but no handover between sectors has been considered. Table 2 summarizes some other important parameters concerning the system level simulation tool.

### *C. Link-to-System Level Interfaces*

Link Adaptation algorithms select the optimum transport mode based on the experienced channel quality conditions. As a result, the representation of such conditions is a key aspect to be considered whenever analyzing the performance and operation of LA. In order to reduce the complexity of system level simulations, the effects at the physical layer are generally included by means of simple Look-Up Tables (LUTs). Since the work reported in [8] demonstrated the importance of using accurate LUTs for the study of adaptive radio link techniques, an advanced link-to-system level interface, working at the burst level, has been considered in this study. This interface is composed of two LUTs. The interface requires as input from the system level the mean Carrier to Interference Ratio (CIR)

TABLE II. SIMULATION PARAMETERS

| Parameter | Value |
|---|---|
| Cluster size | 4 |
| Cell radius | 1km |
| Sectorisation | 120º |
| Modelled interference | 1st and 2nd co-channel tiers |
| Nº of modelled cells (wrap-around) | 25 |
| Slots per sector | 16 |
| Traffic type and load | H.263 video: 6 users/sector<br>WWW: 3 users/sector<br>Email: 3 users/sector |
| Pathloss model | Okumura-Hata |
| Shadowing | Log-normal distribution<br>6dB standard deviation and a 20m decorrelation distance |
| Vehicular speed | 50km/h |
| ARQ protocol | Only for WWW and email users. Assumed: perfect feedback of ARQ report and no RLC block losses |
| ARQ window size | 64 RLC blocks |
| ARQ report polling period | 16 RLC blocks |

experienced in a given burst. LUT-1 extracts the burst link quality, represented by means of the Bit Error Rate (BER), for the measured burst CIR. As illustrated in Fig. 1, LUT-1 represents a cumulative distribution function (cdf) of the BER for a given CIR. A random process is then used to generate the actual BER from the corresponding cdf (there is a BER cdf for each local mean CIR). The interest of this procedure is to model the effect of fast fading on the BER through a random process, thereby including the fast fading at the system level. The BER is then estimated for the four bursts used to transmit a RLC block and LUT-2 maps the mean BER and the standard deviation of the BER over the four bursts to a corresponding BLER value. Fig. 2 shows LUT-2 for the coding scheme CS1 and a vehicular speed of 50km/h.

### *D. Traffic Models and Quality of Service*

Although this study concentrates on real-time H.263 video transmissions, the evaluation environment also implements two best effort services: email and WWW browsing. Even if results are collected individually for each traffic type, no channel partition between the different services has been applied. The WWW and email traffic sources have been implemented as ON/OFF models [7].

Since the transmission rates offered by the GPRS standard are quite low, this work has considered a H.263 video traffic source with a 16kbit/s mean bit rate. The model employs three different frame types, namely I, P and PB. Each frame type exhibits different statistical properties [9]. The video traffic model considers the following variables: frame size, frame duration, correlation between frame size and frame duration, and frame transition rate. The modeling is performed at two levels. The first one determines the frame type to be generated.

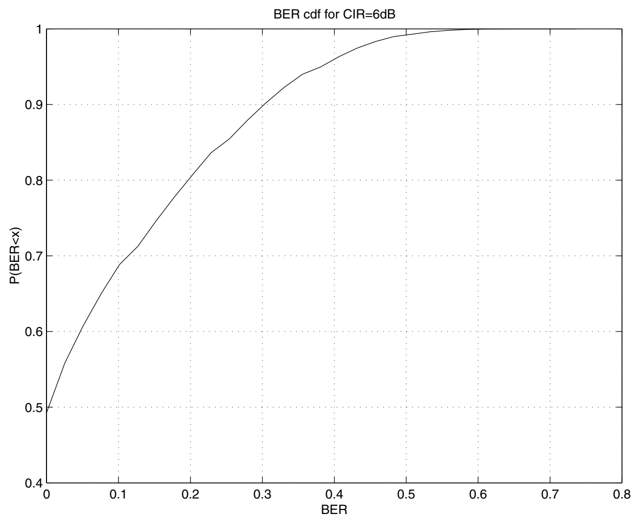


Figure 1.LUT-1 (CIR=6dB, user speed=50km/h)

While I frames are generated at regular intervals, the generation of P and PB frames is determined by means of a Markov Chain. Once the type of frame has been established, the model determines its frame size and duration.

The quality of the received video frames is evaluated by means of the Block Error Rate (BLER). Since [10] has reported that a BLER below 5% would not produce a noticeable video degradation for H.263 transmissions, an error rate target of 5% has been considered in this work.

## III. LINK ADAPTATION ALGORITHMS AND PERFORMANCE EVALUATION

### A. Throughput-based and error-based LA algorithms

The basis of LA is to adaptively select the optimum CS according to the channel quality conditions and a predefined criteria. Since the aim of the throughput-based LA algorithm is to maximize the system throughput, the algorithm will select the CS that maximizes the throughput defined as:

$$\text{Throughput} = R_{CS} \times (1 - BLER_{CS}) \qquad (1)$$

$R_{CS}$ and $BLER_{CS}$ are the data rate and BLER for a given CS.

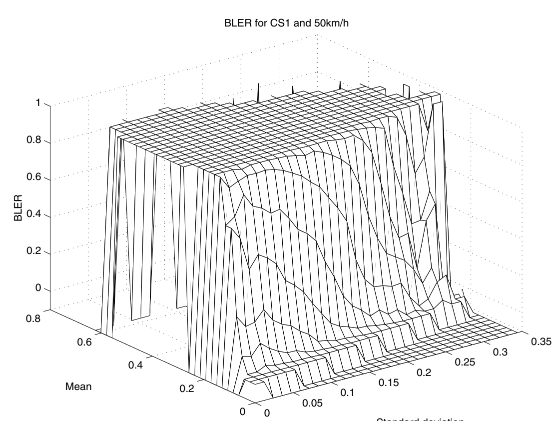


Figure 2.LUT-2 (CS1, user speed=50km/h)

On the other hand, the criteria used by the error-based LA algorithm to select the optimum CS is to achieve the target error rate required by a particular service. For establishing the thresholds defining the boundaries between the regions where each CS is regarded as optimum, the algorithm proposed in [6] uses simple LUTs directly mapping the BLER to the CIR. The thresholds defining the operating regions of each CS were obtained by looking below which CIR a given CS is not able to guarantee a specific BLER performance.

Table 3 illustrates the performance obtained with the throughput-based and error-based LA algorithms. The results shown in this table correspond to the average throughput, the minimum throughput experienced by 95% of the samples (this parameter has been extracted from the system throughput cdf), the average BLER and the percentage of video frames that have been transmitted with a BLER below the established target error rate (5%). Table 3 shows that while the throughput-based LA algorithm obtains a higher throughput performance, the error-based algorithm improves the BLER performance. However, it can be observed that the difference between both algorithms is not very significant, and more importantly, that the achieved average BLER by the error-based LA algorithm is well above the desired target. This pattern was also observed in [6], although the difference between the measured BLER and the target error rate was smaller. In fact, while this difference is above 100% for the results reported in Table 3, the difference was equal to 53% in [6]. The higher difference obtained in our work could be due to the fact that while [6] has been conducted using a simple LUT, this work is based on more sophisticated link-to-system level interfaces. At this point, it is important to note that [8] has already demonstrated the effect that link-to-system level interfaces have on the predicted performance and operation of Link Adaptation.

### B. Proposed LA algorithm

Since the error-based LA algorithm does not achieve the target error rate for H.263 video transmissions, the first objective of this work has been the definition of a simple LA algorithm that can achieve an average BLER below 5%.

Before describing the proposed LA algorithm, it is important to analyze the performance obtained if no LA is used (i.e. a fixed CS is employed during the entire simulation). Such performance, illustrated in Table 4, is useful to explain why the error-based LA algorithm does not perform as expected. As it can be observed from this table, the coding scheme that provides the higher BLER and the lower percentage of video frames transmitted with a BLER lower than 5% is CS4. In fact, all other coding schemes offer an average BLER that satisfies

TABLE III. THROUGHPUT AND ERROR BASED LA ALGORITHMS

| | **Throughput algorithm** | **Error algorithm** |
|---|---|---|
| Mean throughput (kbits/s) | 16.56 | 16.24 |
| Minimum throughput for 95% of samples (kbits/s) | 10.46 | 9.92 |
| Average BLER (%) | 11.18 | 10.60 |
| % of frames with BLER ≤ 5% | 60.75 | 61.10 |

the target error rate for H.263 video transmissions. Since CS4 was the most used coding scheme when considering the application of the error-based LA algorithm (its usage percentage was equal to 57.7%), it can be concluded that the use of CS4 is at the origin of this algorithm's poor BLER performance.

Taking into account the results shown in Table 4, a simple LA algorithm to achieve a low target error rate simply consists in forbidding the use of the less robust coding schemes. The performance obtained with a LA algorithm that forbids the use of CS4 and only considers CS1, CS2 and CS3 in its adaptation process is illustrated in Table 5. As it can observed from this table, the proposed algorithm obtains an average BLER below 5% and transmits the vast majority of video frames with a BLER below 5%. This improvement in terms of BLER performance is due to an important decrease in the proportion of wrong side failures (a wrong side failure corresponds to the case where a user is using a non-optimal CS that is not robust enough for correct reception). In fact, while this proportion was equal to 10.65% in the case of the error-based LA algorithm, it is equal to 5.43% with the proposed LA algorithm. As illustrated in Table 5, the BLER performance achieved with the proposed LA algorithm is obtained at the cost of an important reduction of the throughput and of the proportion of RLC blocks received with the optimal CS. While this proportion was equal to 64.2% with the error-based LA algorithm, it is now equal to 14.2%. However, the proposed algorithm also reduces the signaling load associated with the use of LA; such signaling load can be estimated by means of the average number of CS changes per second. While this number was equal to 5.02 with the error-based LA algorithm, it has decreased to 1.96 with the proposed algorithm. In fact, this reduction in the average number of CS changes per second shows that a high proportion of the signaling load associated with the use of LA is due to the use of the less robust coding scheme (CS4).

As it can be observed from Table 5, it would be possible to further improve the BLER performance if CS3 was also forbidden in the LA process. Once more this improvement in terms of BLER would be obtained at the cost of a poorer throughput performance.

TABLE IV. PERFORMANCE WITH FIXED CODING SCHEMES

| | CS1 | CS2 | CS3 | CS4 |
|---|---|---|---|---|
| Average BLER (%) | 1.08 | 2.54 | 4.25 | 16.93 |
| % of frames with BLER ≤ 5% | 96.42 | 92.12 | 87.08 | 59.13 |

TABLE V. PROPOSED LA ALGORITHMS

| | Without CS4 | Without CS4 and CS3 |
|---|---|---|
| Mean throughput (kbits/s) | 14.30 | 12.92 |
| Minimum throughput for 95& of samples (kbits/s) | 10.29 | 10.63 |
| Average BLER (%) | 4.57 | 2.19 |
| % of frames with BLER ≤ 5% | 83.20 | 91.41 |

### C. *Trade-off LA algorithm*

This section presents and evaluates another LA algorithm that controls the experienced error rate but also offers the possibility to trade-off between throughput and BLER. This proposal could actually allow operators to dynamically adjust the operation and performance of LA according to its needs and to the expected QoS required by the services it offers.

The proposed trade-off LA algorithm regularly evaluates the experienced average BLER during previous transmissions. Consequently, one important parameter in the design of this algorithm will be how often the average BLER is estimated; this parameter is referred in the rest of this paper as averaging period. If the estimated average BLER is above the specific target error rate (5% in this work), the LA algorithm will forbid, for the next averaging period, the use of the less robust CS out of all CS that could be used during the previous averaging period. On the other hand, if the estimated average BLER is below 5%, the LA algorithm will allow the use of the more robust CS out of all CS that were not permitted during the previous averaging period. Obviously, if the average BLER is below 5% and all CS are permitted, the proposed LA algorithm will take no action and its operation will be maintained. Also, if the average BLER is above 5% and only the use of CS1 was allowed during the previous averaging period, the proposed LA algorithm will maintain CS1 as the only possible CS during the next averaging period. The performance achieved with the proposed LA algorithm and different averaging periods (in seconds) is illustrated in Tables 6 and 7. Table 7 shows the usage percentage of each CS, the proportion of RLC blocks received with the optimal CS and the proportion of wrong and right side failures (a right-side failure corresponds to the case where a user is using a non-optimal CS but one robust enough for correct reception). Table 6 shows that the proposed algorithm manages to obtain a low BLER and also to trade-off between throughput and error rate performances by simply changing the averaging period. In fact, the results obtained show that the BLER can be decreased, until a certain level, by increasing the averaging period. As shown in Table 7, increasing the averaging period reduces the use of CS4 and the proportion of wrong side failures and increases the proportion of right side failures. While the use of CS4 and an important proportion of wrong side failures are generally at the origin of transmission errors, a high proportion of right side failures indicates that the proposed algorithm tends to protect, sometimes in excess, the transmissions from radio propagation errors. However, this effect is not maintained if the averaging period is increased in excess (i.e. when considering an averaging period of 36 sec). As the results presented in Table 7 demonstrate, using an averaging period of 36 seconds provides a higher proportion of wrong side failures than using any of the other averaging periods. This increase is not due to a higher usage of CS4 but to an important increase in the usage percentage of CS3, a coding scheme with a low error protection, and a decrease in the usage percentage of the more robust coding scheme (CS1).

It is possible to further reduce the experienced BLER by considering the use of hysteresis thresholds. In this case, the LA algorithm operates in the same way as when not considering hysteresis thresholds if the average BLER

TABLE VI. TRADE-OFF LA ALGORITHMS: PERFORMANCE

| | 1 sec | 12 sec | 18 sec | 36 sec |
|---|---|---|---|---|
| Mean throughput (kbits/s) | 15.31 | 15.02 | 14.79 | 14.41 |
| Minimum throughput for 95% of samples (kbits/s) | 9.24 | 9.04 | 9.04 | 9.30 |
| Average BLER (%) | 6.35 | 5.65 | 5.61 | 6.13 |
| % of frames with BLER ≤ 5% | 75.10 | 77.87 | 78.11 | 76.70 |

TABLE VII. TRADE-OFF LA ALGORITHMS: OPERATION

| | 1 sec | 12 sec | 18 sec | 36 sec |
|---|---|---|---|---|
| Usage % of CS1 | 14.23 | 13.07 | 12.93 | 10.94 |
| Usage % of CS2 | 16.09 | 22.49 | 24.48 | 26.50 |
| Usage % of CS3 | 35.65 | 36.54 | 38.19 | 45.12 |
| Usage % of CS4 | 34.03 | 27.90 | 24.40 | 17.44 |
| Optimal CS (%) | 43.38 | 37.03 | 33.70 | 27.81 |
| Wrong side failures (%) | 6.41 | 5.99 | 5.96 | 6.47 |
| Right side failures (%) | 50.21 | 56.99 | 60.34 | 65.72 |

measured during the previous averaging period is higher than 5%. On the other hand, the algorithm allows the use of the most robust CS out of all CS that were previously forbidden, only if the averaged BLER goes below 4% and not 5%. The operation of the proposed algorithm has also been evaluated with hysteresis thresholds equal to a BLER of 4 and 4.5%. In this case, the performance will be shown for an averaging period of 14 seconds and not 18 since a lower BLER (5.2%) was measured with the 14 seconds period. As shown in Fig. 3, the use of hysteresis thresholds reduces the BLER performance and increases the percentage of video frames transmitted with a BLER below 5%. Fig. 3 also demonstrates that the best BLER performance is obtained with the more restrictive hysteresis thresholds.

## IV. CONCLUSIONS

The design of Link Adaptation algorithms has mainly concentrated on improving the throughput performance. While this approach can be appropriate for best-effort services, multimedia services have important requirements in terms of transmission delay and error rate performance. In this context, this paper proposes and evaluates different simple LA algorithms designed to improve the error rate performance. In particular, this work has focused on real-time H.263 video transmissions in a GPRS-like network. The results obtained demonstrate that the proposed algorithms are able to maintain a low error rate, at the cost of reducing the throughput performance. The study conducted has also shown that one of

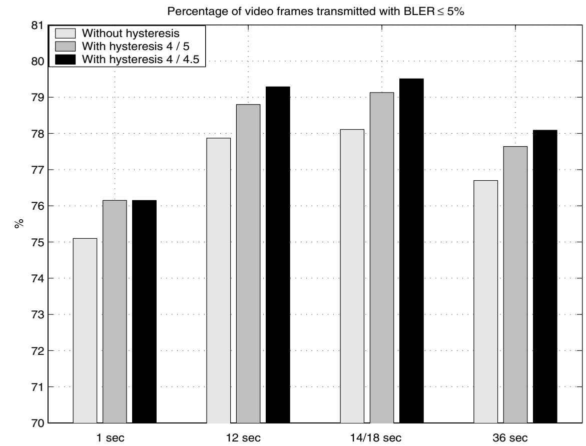


Figure 3. Effect of hysteresis thresholds on the BLER performance

the proposed algorithms is able to trade-off between throughput and BLER by simply modifying one of its operating parameters. This feature could allow operators to dynamically configure the LA algorithm according to the QoS required by the particular service it is providing at a given moment in time.

## REFERENCES


[1] J. Dunlop, J. Pons, J. Gozalvez and P. Atherton," Real-time GSM Link Adaptation hardware demonstrator", Proc. of the IEEE VTC-Spring 2000, pp 590-594, May 2000, Tokyo.

[2] ETSI-SMG, "EDGE Feasibility study, Work item 184; Improved Data Rates through Optimized Modulation," Tdoc 97-331, December 1997.

[3] C. Kodikara, S.N Fabri and A.M. Kondoz, "Performance improvement for real-time video communications by Link Adaptation in EGPRS networks", Proc. of the IEE 3G Mobile Communication Technologies Conference, pp 489-494, May 2002, London.

[4] W. Luo, K. Balachandran, S. Nanda and K. Chang, "Packet size dependent Link Adaptation for wireless packet data", Proc. of IEEE Globecom, pp 53-56, November 2000, San Francisco.

[5] J. Gozalvez, M. López-Benítez, O. Lázaro and J. Evangelio, "Design and evaluation of a Link Adaptation algorithm for enhanced video service provision within mobile radio environments", to be published at Proc. of the IEEE PIMRC2004, September 2004, Barcelona.

[6] K.K Leung, P.F. Driessen, K. Chawla and X. Qiu, "Link Adaptation and power control for streaming services in EGPRS wireless networks", IEEE Journal on Selected Areas in Communications, vol. 19, nº10, pp 2029-2039, October 2001.

[7] J. Gozalvez and J. Dunlop, "On the dynamics of Link Adaptation updating periods for packet switched systems", Proc. of the 4th WPMC, pp 609-614, September 2001, Aalborg.

[8] J. Gozalvez and J. Dunlop, "Link level modelling techniques for analysing the configuration of Link Adaptation algorithms in mobile radio networks", Proc. of European Wireless 2004, pp 325-330, February 2004, Barcelona.

[9] O. Lazaro, D. Girma and J. Dunlop, "H.263 video traffic modelling for low bit rate wireless communications", to be published at Proc. of the IEEE PIMRC2004, September 2004, Barcelona.

[10] L. Hanzo, P. Cherriman and J. Streit, "Wireless video communications: second to third generation systems and beyond", IEEE Press, 2001.