

EXHIBIT 3

IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

NOKIA CORPORATION, Plaintiff, v. APPLE INC., Defendant.	))))) C.A. No. ____________________)) JURY TRIAL DEMANDED))

COMPLAINT FOR PATENT INFRINGEMENT
AND DECLARATORY JUDGMENT

Plaintiff Nokia Corporation ("Nokia"), on personal knowledge as to its own acts, and on information and belief as to all others based on its investigation, alleges as follows:

INTRODUCTION

1. This is an action brought by Nokia against Apple Inc. ("Apple") for Apple's infringement of Nokia's patents. In particular, Nokia seeks remedies for Apple's infringement of Nokia's U.S. Patent Nos. 5,802,465 ("the 465 Patent"), 5,862,178 ("the 178 Patent"), 5,946,651 ("the 651 Patent"), 6,359,904 ("the 904 Patent"), 6,694,135 ("the 135 Patent"), 6,775,548 ("the 548 Patent"), 6,882,727 ("the 727 Patent"), 7,009,940 ("the 940 Patent"), 7,092,672 ("the 672 Patent"), and 7,403,621 ("the 621 Patent") (collectively, "the patents-in-suit").

2. Each of the patents-in-suit is essential to one or more of the following standards: the Global System for Mobile Communications ("GSM") Standard, the Universal Mobile Telecommunications System ("UMTS") Standard, and the Institute of Electrical and Electronic Engineers ("IEEE") 802.11 Standard.

3. Nokia has declared each of the patents-in-suit as essential to the GSM, UMTS, and/or 802.11 Standards, where applicable, and undertaken -- in accordance with the

applicable rules of the standard setting organizations ("SSO") -- to grant licenses under each of the patents-in-suit on fair, reasonable, and nondiscriminatory ("FRAND") terms and conditions (in some cases, alternatively referred to as "reasonable and non-discriminatory," or "RAND," terms).

4. On the basis of Nokia's licensing commitments, Apple has the right to be granted license(s) under F/RAND terms and conditions with respect to a Standard.

5. Prior to filing this Complaint, Nokia has made various offers to Apple for the F/RAND terms and conditions of a license agreement under which each of the patents-in-suit could be licensed either individually or together with other Nokia essential patents (i.e., a portfolio license). In its offers to Apple, Nokia has specified both a portfolio rate and an average per-patent royalty rate which Apple could have accepted within a reasonable time for each of the patents-in-suit.

6. Apple has rejected Nokia's offers for the F/RAND terms and conditions both on a portfolio and on a per-patent basis and thereby refused to compensate Nokia on F/RAND terms for its use of Nokia's patented technologies, including each of the patents-in-suit.

7. In order to be fairly and adequately rewarded for the use of Nokia patented technology in the implementation of the standards, Nokia seeks by this action F/RAND compensation for Apple's use of the patents-in-suit. In addition, Nokia seeks a declaration (i) that the patents-in-suit are infringed by Apple's products complying with the respective Standards and that the patents-in-suit are not invalid or unenforceable (ii) that Nokia has complied with its obligations under the F/RAND undertakings by negotiating in good faith and offering and specifying F/RAND terms and conditions for the patents-in-suit, (iii) that Apple has refused to compensate Nokia on F/RAND terms for the patents-in-suit in breach of its obligation

to pay for the use of the Nokia patents, and (iv) that Nokia is entitled to an injunction until and unless Apple pays F/RAND compensation, together with interest, for past infringement of the patents-in-suit and irrevocably commits to pay such compensation in the future.

PARTIES

8. Plaintiff Nokia is incorporated under the laws of Finland and has its principal place of business at Keilalahdentie 4, Espoo, Finland.

9. Nokia was founded in 1865 and is the world's largest manufacturer of mobile telephones. Nokia is one of the champions of wireless cellular communications and has received numerous awards and accolades for its achievements, including introducing the first car phone on the first international cellular mobile network in 1981.

10. Nokia's innovations continue today. In 1991, the world's first genuine call on GSM was made with a Nokia phone. In 1996, Nokia introduced the Nokia 9000 Communicator, which was the first all-in-one phone, fax, calendar, e-mail and Internet device in a hand-portable size. The Nokia 8110i, introduced in 1997, was the first mobile phone with a dynamic menu supporting Smart Messaging. Just two years later, Nokia introduced the Nokia 7110, which was the first mobile phone compliant with the Wireless Application Protocol 1.1, which provided access to mobile Internet services, such as banking, e-mail, and news, as well as the first phone with predictive text input.

11. In 2001, Nokia made the world's first 3G WCDMA voice call on a commercial system, and launched its first imaging phone with an integrated camera, the Nokia 7650. In 2002, Nokia introduced the world's first UMTS/GSM dual mode phone, and the first Nokia phone to record video simultaneously with sound. The Nokia 5140, launched in 2003 was

the first Push-to-Talk GSM handset. In 2006, Nokia introduced the N95, which was the first Nokia phone with built-in GPS.

12. Research is one of the keys to Nokia's success. As of December 2008, Nokia had research and development presence in 16 countries and employed over 39,000 people in research and development. Such research and development led to the innovations found in the patents-in-suit.

13. In the 1980s, Nokia led the charge to establish the communications protocols that are still used today. Without Nokia's contributions and innovation, the world would not have the communications standards that it has today. Nokia continues to be a leader in mobile communications worldwide and continues to invest millions of dollars annually in new developments in mobile communications.

14. Upon information and belief, Defendant Apple is a corporation duly organized and existing under the laws of the state of California and has a principal place of business at 1 Infinite Loop, Cupertino, California 95014.

15. Upon information and belief, Apple did not make telephones, much less mobile telephones, until 2007. Apple's wireless communication devices take advantage of the decades of continued investments by Nokia to build today's communication protocols. By refusing to compensate Nokia for its patented technologies, Apple is attempting to get a "free-ride" on the billions of dollars that Nokia has invested in research and development to provide the public with the wireless communications it enjoys today.

JURISDICTION AND VENUE

16. This is an action arising under the patent laws of the United States. Accordingly, this Court has subject matter jurisdiction pursuant to 28 U.S.C. §§ 1331 and 1338(a).

17. This Court has personal jurisdiction over Apple because Apple has established minimum contacts with the forum. Apple manufactures (directly or indirectly through third party manufacturers)and/or assembles products that are and have been used, offered for sale, sold, and purchased in Delaware. Apple, directly and/or through its distribution network, places wireless communication devices within the stream of commerce, which stream is directed at this district, with the knowledge and/or understanding that such products will be sold in the State of Delaware. Therefore, the exercise of jurisdiction over Apple would not offend traditional notions of fair play and substantial justice.

18. Apple does business in this district, including providing products that are used, offered for sale, sold, and have been purchased in Delaware. Venue is proper in this district pursuant to 28 U.S.C. §§ 1391(b), (c), (d) and 1400(b).

FACTUAL BACKGROUND

The Mobile Wireless Industry

19. The wireless devices developed and marketed by Nokia and Apple connect to a variety of wireless networks, including the networks of wireless carriers to provide telecommunications service. Carriers operate wireless systems that enable users to place and receive telephone calls, send and receive e-mails, and connect to the internet through wireless devices. Leading carriers in the United States include AT&T (formerly Cingular), T-Mobile USA, Verizon Wireless, and Sprint.

20. Companies around the world manufacture wireless devices. These manufacturers typically sell their phones to the mobile wireless carriers, which in turn sell the phones to users. Wireless devices contain, among other components, one or more computer chipsets that enable the phone to communicate with the carriers' wireless systems. Carriers, device manufacturers, and chipset manufacturers must create equipment and devices compatible with each other by using common mobile wireless technology. Since carriers, device manufacturers, and chipset manufacturers must create equipment and devices compatible with each other to provide mobile wireless services, developers and manufacturers participate in the crucial process of standards development.

21. The progression from cell phones, which primarily focus on voice communications, to smart phones required more advanced mobile wireless technologies for communications involving transmission of data such as e-mail. Since the mass market introduction of the cell phone in the 1980s, mobile wireless technology has evolved to keep pace with the rising volume of voice traffic as well as to incorporate the data transmission capabilities necessary to support increasingly sophisticated phones and other handheld devices. The technology has evolved in what are commonly referred to as "generations" of mobile wireless technology.

22. The first generation of mobile wireless technology (1G) consisted of analog devices and networks that carried only voice traffic. The second generation of mobile wireless technology (2G) began the transition to digital devices and networks providing more efficient use of available spectrum for voice traffic and limited support for data-intensive applications such as paging and text messaging. The emergence of 2G technologies coincided with the growing commercial use of the Internet. The greater data capacity of advanced 2G

networks allowed for the development of the first smart phones, which offered new capabilities such as taking and transmitting photographs, sending and receiving email, and limited web browsing. Third generation (3G) wireless technology supports more advanced data intensive services, such as multimedia, web browsing, music and video downloads, e-commerce, and position location. Fourth Generation (4G) wireless technology is currently being developed. 4G technologies will provide voice, data, and streamed media at much higher data rates compared to the previous generations. Almost all wireless carriers currently support and provide 2G technology, and most have also introduced 3G networks and services. Some carriers have announced plans for migration to 4G networks and services in the coming years.

The Importance of Standards

23. The UMTS and GSM standards, as well as other mobile radio standards, were developed under the patronage of the European Telecommunications Standards Institute ("ETSI"). ETSI is a non-profit institution that was founded in 1988 through an initiative of the European Commission by several companies active in mobile communication with the objective to develop a common mobile radio standard for Europe. Since it was founded, ETSI has grown to include approximately 700 members from 56 countries. Among these members are virtually every company active in the mobile radio sector, who together account for a substantial share of the supply of mobile telecommunications equipment and services. Nokia and Apple are both members of ETSI.

24. ETSI brings important market participants in the mobile radio sector together. Within the context of ETSI, the members develop technical standards, which often lead to a factually binding industry standard. In some cases, national or international regulatory bodies require adherence to particular ETSI standards.

25. Many ETSI members, including Nokia, are engaged in research and development of new telecommunications technologies, and own intellectual property rights relating to different elements of such technologies. Accordingly, when ETSI adopts technical standards, it must take into account that many elements of the standards are likely to be covered by such intellectual property rights. Therefore, others wishing to exploit the standard may need licenses for the essential intellectual property rights to do so. ETSI has therefore adopted an Intellectual Property Policy ("the ETSI IPR Policy") to govern the manner in which ETSI will take account of such intellectual property rights in the process leading to the adoption of ETSI standards.

26. The ETSI IPR policy was adopted in 1994 and the policy has been part of the "ETSI Directives" since December 2004. Its provisions are further explained in the ETSI Guide on Intellectual Property Rights.

27. The objectives of the ETSI IPR Policy are defined in its Clause 3. Clause 3.1 provides as follows:

> It is ETSI's objective to create STANDARDS and TECHNICAL SPECIFICATIONS that are based on solutions which best meet the technical objectives of the European telecommunications sector, as defined by the General Assembly. In order to further this objective the ETSI IPR POLICY seeks to reduce the risk to ETSI, MEMBERS, and others applying ETSI STANDARDS and TECHNICAL SPECIFICATIONS, that investment in the preparation, adoption and application of STANDARDS could be wasted as a result of an ESSENTIAL IPR for a STANDARD or TECHNICAL SPECIFICATION being unavailable. In achieving this objective, the ETSI IPR POLICY seeks a balance between the needs of standardization for public use in the field of telecommunications and the rights of the owners of IPRs.

28. In order to achieve its objectives, the ETSI IPR Policy contains rules regarding the disclosure of essential IPR and rules regarding their licensing on FRAND terms. Members are obligated to use their reasonable endeavors to inform ETSI of essential IPRs in a timely manner, and voluntarily undertake to grant licenses on FRAND terms and conditions.

Therefore, ETSI allows its members to hold and benefit from any IPRs which they may own, including the right to refuse the granting of licenses.

29. Clause 6.1 of the ETSI IPR Policy provides:

> When an essential IPR relating to a particular standard or technical specification is brought to the attention of ETSI, the director-general of ETSI shall immediately request the owner to give within three months an irrevocable undertaking in writing that it is prepared to grant irrevocable licenses on fair, reasonable, and non-discriminatory terms and conditions under such IPR to at least the following extent . . .

30. The ETSI IPR Policy provides that firms owning potentially essential patents will provide undertakings of the kind envisaged by clause 6.1 of the ETSI IPR Policy preferably before adoption of the respective standard. If an owner of an essential IPR does not submit this declaration, keeping its technology proprietary, alternatives are sought to the essential technology, which would not require the infringement of the IPR, pursuant to Clause 8.1.1 of the ETSI IPR Policy. If no technical alternative is available, the development of the respective standards is ceased, under Clause 8.1.2 of the ETSI IPR Policy.

31. Pursuant to the ETSI IPR Policy, Nokia has submitted declarations for certain of the patents-in-suit. For example, with respect to the 465 Patent, Nokia submitted a declaration stating the following:

> The signatory has notified ETSI that it is the proprietor of the IPRs listed above and has informed ETSI that it believes that the IPRs may be considered ESSENTIAL to the Standards listed above. The SIGNATORY and/or its AFFILIATES hereby declare that they are prepared to grant irrevocable licences under the IPRs on terms and conditions which are in accordance with Clause 6.1 of the ETSI IPR Policy, in respect of the STANDARD, to the extent that the IPRs remain essential. ...
>
> The construction, validity and performance of this DECLARATION shall be governed by the laws of France.

32. Like ETSI, the Institute of Electrical and Electronics Standards Association (IEEE-SA) is a developer of industry standards in a number of industries, including

telecommunications, information technology, nanotechnology, and information assurance. Among the standards developed by IEEE-SA is IEEE 802.11, the standard for WLAN and IEEE 802.16, the standard for WiMax.

33. Like ETSI, many IEEE-SA members, including Nokia, are engaged in the research and development of new technologies and own intellectual property rights relating to different elements of such technologies. Accordingly, IEEE-SA has adopted a similar intellectual property policy as ETSI in the IEEE-SA Standards Board Bylaws ("IEEE-SA Bylaws").

34. Clause 6.2 of the IEEE-SA Bylaws states that when a standard includes the use of Essential Patent Claims, a "letter of assurance" with regard to the essential patent may be requested. That letter of assurance may include:

> . . . .
>
> A statement that a license for a compliant implementation of the standard will be made available to an unrestricted number of applicants on a worldwide basis without compensation or under reasonable rates, with reasonable terms and conditions that are demonstrably free of any unfair discrimination. At its sole option, the Submitter may provide with its assurance any of the following: (i) a not-to-exceed license fee or rate commitment, (ii) a sample license agreement, or (iii) one or more material licensing terms.

35. Clause 6.2 of the IEEE-SA Bylaws further provides for an instance where a party providing a letter of assurance discovers additional claims that are essential to a standard:

> If, after providing a Letter of Assurance to the IEEE, the Submitter becomes aware of additional Patent Claim(s) not already covered by an existing Letter of Assurance that are owned, controlled, or licensable by the Submitter that may be or become Essential Patent Claim(s) for the same IEEE Standard but are not the subject of an existing Letter of Assurance, then such Submitter shall submit a Letter of Assurance stating its position regarding enforcement or licensing of such Patent Claims. For the purposes of this commitment, the Submitter is deemed to be aware if any of the following individuals who are from, employed by, or otherwise represent the Submitter have personal knowledge of additional potential Essential Patent Claims, owned or controlled by the Submitter, related to a [Proposed] IEEE Standard and not already the subject of a previously submitted

> Letter of Assurance: (a) past or present participants in the development of the [Proposed] IEEE Standard, or (b) the individual executing the previously submitted Letter of Assurance.

36. Clause 6.2 of the IEEE-SA Bylaws also provides that a letter of assurance, once submitted, is irrevocable.

37. Pursuant to the IEEE-SA Bylaws, Nokia has submitted letters of assurance for certain of the patents-in-suit. For example, with respect to the 465 Patent, Nokia submitted a letter of assurance stating the following:

> In accordance with Clause 6.2 of the *IEEE-SA Standards Board Bylaws*, the Submitter hereby declares the following: …
>
> The Submitter may own, control or have the right to license Patent Claims that might be or become Essential Patent Claims. With respect to such Essential Patent Claims, the submitter's licensing position is as follows: …
>
> The Submitter will grant a license under reasonable rates to an unrestricted number of applicants on a worldwide basis with reasonable terms and conditions that are demonstrably free of unfair discrimination.

F/RAND

38. Standards Setting Organizations ("SSOs") are formed to allow wide promulgation and utilization of commonly defined standards. These standards must be available and accessible in order to produce the intended efficiency gains and benefits and thereby for the standardization process itself to comply with competition law. Intellectual Property Rights policies ("IPR Policies"), like those described above, provide essential IPR holders committing to license on F/RAND terms with the benefit of collecting F/RAND compensation from a far larger market than they would have enjoyed if the protected technology had not been incorporated in the standard. Because competing proprietary technologies and systems have been abandoned in favor of a single, universal, and standardized system and set of technologies, a holder of an essential IPR can collect royalties on a large volume of standards-compliant

products from a wide variety of manufacturers worldwide. In contrast, if the IPR holder's protected technology was only used in one of a number of competing systems or proprietary technologies, the patent holder could only generate returns on its R&D investments through differentiation and -- if it chose to license -- only collect royalties from manufacturers who chose to market and sell products for the narrow proprietary technology. This is why committing to F/RAND licensing is advantageous and rarely refused by essential IPR holders.

39. An IPR holder that has voluntarily undertaken to license its IPRs on F/RAND terms (instead of keeping the inventions proprietary) has irrevocably committed to allow the standard to be implemented under its IPR on F/RAND basis and thereby waived -- absent exceptional circumstances -- its legally defined right to exclude others from practicing the standard under its IPR. This also means that the IPR holder cannot use its hold-up power resulting from the incorporation of its technology into the standard and the IPR holder's right to exclude to extort royalties that do not comply with F/RAND.

40. Once an IPR holder has made a F/RAND commitment, all manufacturers have the right to implement the standard in their products and use the inventions from any declared essential IPRs. There is no need to wait until all the particular F/RAND terms and conditions have been negotiated with the IPR holder or until a definitive license agreement is executed setting out those terms. However, it is clear that in return for the right to practice the standard under the essential IPRs, implementing manufacturers have the obligation to pay F/RAND compensation for the IPR used (to the extent not invalid or unenforceable). For example, according to the ETSI IPR Policy Clause 3.2:

> IPR holders whether members of ETSI and their AFFILIATES or third parties, should be adequately and fairly rewarded for the use of their IPRs in the implementation of STANDARDS and TECHNICAL SPECIFICATIONS.

41. Save for cases where the manufacturer refuses to take a license altogether, it follows from F/RAND licensing commitments that the IPR holder has a duty to negotiate in good faith and propose F/RAND terms. Negotiations over F/RAND terms may cover the essential IPR portfolio as a whole but, if requested, F/RAND terms should be available for each patent separately.

42. If the implementer refuses to take a license altogether or refuses to pay F/RAND compensation for valid and enforceable IPRs used by it, exceptional circumstances are present and the IPR holder may seek an injunction to prevent the implementer from continuing to manufacture standard-compliant products without payment. The injunction only extends for so long as the manufacturer refuses to pay F/RAND compensation.

Apple's Refusal to pay F/RAND Compensation

43. Nokia has irrevocably undertaken the obligation to grant license(s) on F/RAND terms and conditions to its essential patents, including the patents-in-suit, and Apple has the corresponding right to claim licenses on F/RAND terms on the basis of Nokia's undertakings.

44. In compliance with its declarations and undertakings which Nokia submitted with regard to the patents-in-suit, prior to filing this complaint Nokia has negotiated in good faith over the F/RAND licensing terms with Apple. Nokia has made various offers to Apple for the F/RAND terms and conditions of a license agreement under which the patents-in-suit could be licensed either individually or in combination with other Nokia essential patents. In its offers, made subject to reciprocity, Nokia has defined both a portfolio rate and an average per patent royalty rate which Apple could have accepted within a reasonable time. Nokia has also provided Apple with information on the method used to calculate royalties as well as claim charts assisting Apple with its technical analysis.

45. Apple has rejected Nokia's offers for the F/RAND terms and conditions both on a portfolio and on a per patent basis and thereby refused to compensate Nokia on F/RAND terms for the use of Nokia patented technology, including the patents-in-suit.

46. Due to Apple's violation of its obligation to pay F/RAND compensation for the use of Nokia's patents, Nokia has no choice but to file this Complaint in order to enforce its right to be compensated on a F/RAND basis for the use of the patents in suit in Apple's standards-compliant products, and to prevent further infringement unless and until Apple pays F/RAND compensation, together with interest, for its past infringement and irrevocably commits to payment of such compensation in the future.

OVERVIEW OF THE PATENTS-IN-SUIT.

47. The patents-in-suit are a reflection of Nokia's research and development and achievements in the world of mobile communications. To provide a few examples, Nokia is a leader in wireless data and owns important patents in this area. Today's wireless devices are used for a wide variety of tasks, such as sending email, browsing the Internet, and downloading applications. These tasks all involve Nokia's advances in wireless data. Without these advances, it would be difficult to work remotely from a coffee shop or download a new game to a phone.

48. Nokia is also a leader in speech coding and owns important patents in this area. In order to send audio, today's phones transmit the audio as a series of 1's and 0's. Speech coding is the backbone of any digital wireless system. Without speech coding, it would be difficult to talk clearly or listen to music without overwhelming the limited resources of the network.

49. Nokia is also a leader in security and encryption and owns important patents in this area. Today's wireless devices are frequently used for e-commerce and other purchases. Nokia's technology allows people to use their wireless devices to conduct business without their confidential information being intercepted.

Wireless Data Patents

50. The 465 Patent, entitled *Data Transmission in a Radio Telephone Network*, was duly and lawfully issued on September 1, 1998. Nokia is the current owner of all rights, title, and interest in the 465 Patent. A true and correct copy of the 465 Patent is attached hereto as Exhibit A.

51. The 465 Patent is essential and has been declared essential to at least the GSM, UMTS, and IEEE 802.11 standards. The 465 Patent invention allows communication over wireless networks while conserving resources on the network. It provides for the formation of a virtual data channel, such that a real data channel can be quickly established when data transmission is desired.

52. The 904 Patent, entitled *Data Transfer in a Mobile Telephone Network*, was duly and lawfully issued on March 19, 2002. Nokia is the current owner of all rights, title, and interest in the 904 Patent. A true and correct copy of the 904 Patent is attached hereto as Exhibit B.

53. The 904 Patent is essential and has been declared essential to at least the GSM and IEEE 802.11 standards. The 904 Patent allows for simpler communication on the networks. The invention provides that, in a radio block to be coded, user data is transferred in octet form to simplify the flow of data in the network.

54. The 135 Patent, entitled *Measurement Report Transmission in a Telecommunications System*, was duly and lawfully issued on February 17, 2004. Nokia is the

current owner of all rights, title, and interest in the 135 Patent. A true and correct copy of the 135 Patent is attached hereto as Exhibit C.

55. The 135 Patent is essential and has been declared essential to at least the GSM standard. The 135 Patent provides an efficient method of communicating information about a mobile device operating in downlink transfer by enabling the mobile device to respond to polling codes with messages that indicate the condition of the mobile device.

56. The 548 Patent, entitled *Access Channel for Reduced Access Delay in a Telecommunications System*, was duly and lawfully issued on August 10, 2004. Nokia is the current owner of all rights, title, and interest in the 548 Patent. A true and correct copy of the 548 Patent is attached hereto as Exhibit D.

57. The 548 Patent is essential and has been declared essential to at least the UMTS standard. The 548 Patent enables a mobile station to access the network with less delay. The '548 invention enables access requests to be adjusted based on channel conditions, reducing overall access delays.

58. The 672 Patent, entitled *Reporting Cell Measurement Results in a Cellular Communication System*, was duly and lawfully issued on August 15, 2006. Nokia is the current owner of all right, title, and interest in the 672 Patent. A true and correct copy of the 672 Patent is attached hereto as Exhibit E.

59. The 672 Patent is essential and has been declared essential to at least the GSM standard. The 672 Patent enables a mobile device to report an increased number of signal quality measurements to a mobile network.

Speech Coding Patents

60. The 178 Patent, entitled *Method and Apparatus for Speech Transmission in a Mobile Communications System*, was duly and lawfully issued on January 19, 1999. Nokia

is the current owner of all rights, title, and interest in the 178 Patent. A true and correct copy of the 178 Patent is attached hereto as Exhibit F.

61. The 178 Patent is essential and has been declared essential to at least the GSM standard. The 178 Patent ensures clear, efficient speech communications over mobile networks. The 178 Patent invention enables multiple speech coding methods to operate at different transmission rates by using two stages of channel encoding, one of which is dependent on the speech coding method, and one of which is not dependent on the speech coding method.

62. The 651 Patent, entitled *Speech Synthesizer Employing Post-Processing for Enhancing the Quality of the Synthesized Speech*, was duly and lawfully issued on August 31, 1999. Nokia is the current owner of all rights, title, and interest in the 651 Patent. A true and correct copy of the 651 Patent is attached hereto as Exhibit G.

63. The 651 Patent is essential and has been declared essential to at least the GSM standard. The 651 Patent ensures clear voice and audio communications over mobile networks. The 651 Patent invention provides for a postfilter for processing speech signals derived from an excitation code book and adaptive code book of a speech decoder.

Security and Encryption Patents

64. The 727 Patent, entitled *Method of Ciphering Data Transmission in a Radio System*, was duly and lawfully issued on April 19, 2005. Nokia is the current owner of all rights, title, and interest in the 727 Patent. A true and correct copy of the 727 Patent is attached hereto as Exhibit H.

65. The 727 Patent is essential and has been declared essential to at least the UMTS standard. The 727 Patent ensures secure transmission of data over mobile networks. The '727 invention prevents data from falling into the wrong hands by using a ciphering algorithm with a channel-specific parameter among its inputs.

66. The 940 Patent, entitled *Integrity Check in a Communication System*, was duly and lawfully issued on March 7, 2006. Nokia is the current owner of all rights, title, and interest in the 940 Patent. A true and correct copy of the 940 Patent is attached hereto as Exhibit I.

67. The 940 Patent is essential and has been declared essential to at least the UMTS standard. The 940 Patent ensures secure transmission of data over mobile networks. The 940 Patent protects communications using an integrity algorithm calculated from values including channel identity information.

68. The 621 Patent, entitled *System for Ensuring Encrypted Communication After Handover*, was duly and lawfully issued on July 22, 2008. Nokia is the current owner of all rights, title, and interest in the 621 Patent. A true and correct copy of the 621 Patent is attached hereto as Exhibit J.

69. The 621 Patent is essential and has been declared essential to at least the GSM and UMTS standards. The 621 Patent ensures continued secure transmissions during a handover by communicating information about encryption algorithms supported by a mobile station between radio access networks.

APPLE'S INFRINGEMENT

70. Upon information and belief, Apple has infringed and continues to infringe each of the patents-in-suit by engaging in acts constituting infringement under 35 U.S.C. § 271, including but not necessarily limited to one or more of making, using, selling and offering to sell, in this District and elsewhere in the United States, and importing into this District and elsewhere in the United States, one or more products and services that comply with the GSM, UMTS, and/or IEEE 802.11 standards, including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS.

HARM TO NOKIA FROM APPLE'S INFRINGEMENT

71. Nokia is harmed by Apple's failure to pay F/RAND compensation for its use of Nokia patented technology in a way that cannot necessarily be compensated for by a payment of a past due F/RAND royalty alone. Apple's failure to pay a F/RAND rate for the use of the patents-in-suit in its products at the time of their sale allows it to charge less for its products because it does not have to recover the costs of development of the technology used in the device. This allows it to obtain market share that it would otherwise not be able to obtain were its products to bear the costs for the patented technology.

72. Nokia's products, in turn, must bear the costs of the development of the technology that allows them to function in compliance with the relevant standards. This puts Nokia in a competitive disadvantage to "free-riders" such as Apple.

73. Even if Apple were to subsequently pay past due F/RAND royalties, it would still enjoy a market share it otherwise would not have but for the period of "free riding." Nokia would likewise lose its portion of the market share for the period of the "free riding." Due to the difficulty in predicting whether, if at all, such market share can be recovered, Nokia's harm cannot be compensated by payment of past due F/RAND royalties alone.

COUNT I
INFRINGEMENT OF U.S. PATENT NO. 5,802,465

74. Nokia incorporates by reference the allegations set forth in Paragraphs 1-73 of this Complaint as though fully set forth herein.

75. Apple has infringed and is infringing the 465 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 465 Patent.

76. Apple is inducing the infringement of the 465 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

77. Apple is contributing to the infringement of the 465 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

78. Apple's infringement of the 465 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

79. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT II
INFRINGEMENT OF U.S. PATENT NO. 5,862,178

80. Nokia incorporates by reference the allegations set forth in Paragraphs 1-79 of this Complaint as though fully set forth herein.

81. Apple has infringed and is infringing the 178 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services that include an encoder and decoder for simultaneous bi-directional voice and/or data communications, including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 178 Patent. Nokia does not allege infringement by Apple based on the making, using, offering for sale, or selling in the United States any product that does not include an encoder and decoder for simultaneous bi-directional voice and/or data communications.

82. Apple is inducing the infringement of the 178 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

83. Apple is contributing to the infringement to the infringement of the 178 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

84. Apple's infringement of the 178 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

85. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT III
INFRINGEMENT OF U.S. PATENT NO. 5,946,651

86. Nokia incorporates by reference the allegations set forth in Paragraphs 1-85 of this Complaint as though fully set forth herein.

87. Apple has infringed and is infringing the 651 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services that include an encoder and decoder for simultaneous bi-directional voice and/or data communications, including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 651 Patent. Nokia does not allege infringement by Apple based on the making, using, offering for sale, or selling in the United States any product that does not include an encoder and decoder for simultaneous bi-directional voice and/or data communications.

88. Apple is inducing the infringement of the 651 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

89. Apple is contributing to the infringement to the infringement of the 651 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

90. Apple's infringement of the 651 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

91. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT IV
INFRINGEMENT OF U.S. PATENT NO. 6,359,904

92. Nokia incorporates by reference the allegations set forth in Paragraphs 1-91 of this Complaint as though fully set forth herein.

93. Apple has infringed and is infringing the 904 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 904 Patent.

94. Apple is inducing the infringement of the 904 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

95. Apple is contributing to the infringement to the infringement of the 904 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

96. Apple's infringement of the 904 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

97. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT V
INFRINGEMENT OF U.S. PATENT NO. 6,694,135

98. Nokia incorporates by reference the allegations set forth in Paragraphs 1-97 of this Complaint as though fully set forth herein.

99. Apple has infringed and is infringing the 135 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 135 Patent.

100. Apple is inducing the infringement of the 135 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

101. Apple is contributing to the infringement to the infringement of the 135 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

102. Apple's infringement of the 135 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

103. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT VI
INFRINGEMENT OF U.S. PATENT NO. 6,775,548

104. Nokia incorporates by reference the allegations set forth in Paragraphs 1-103 of this Complaint as though fully set forth herein.

105. Apple has infringed and is infringing the 548 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 548 Patent.

106. Apple is inducing the infringement of the 548 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

107. Apple is contributing to the infringement to the infringement of the 548 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

108. Apple's infringement of the 548 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

109. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT VII
INFRINGEMENT OF U.S. PATENT NO. 6,882,727

110. Nokia incorporates by reference the allegations set forth in Paragraphs 1-109 of this Complaint as though fully set forth herein.

111. Apple has infringed and is infringing the 727 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 727 Patent.

112. Apple is inducing the infringement of the 727 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

113. Apple is contributing to the infringement to the infringement of the 727 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

114. Apple's infringement of the 727 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

115. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT VIII
INFRINGEMENT OF U.S. PATENT NO. 7,009,940

116. Nokia incorporates by reference the allegations set forth in Paragraphs 1-115 of this Complaint as though fully set forth herein.

117. Apple has infringed and is infringing the 940 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services

including wireless communication devices such as the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 940 Patent.

118. Apple is inducing the infringement of the 940 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

119. Apple is contributing to the infringement to the infringement of the 940 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

120. Apple's infringement of the 940 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

121. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT IX
INFRINGEMENT OF U.S. PATENT NO. 7,092,672

122. Nokia incorporates by reference the allegations set forth in Paragraphs 1-121 of this Complaint as though fully set forth herein.

123. Apple has infringed and is infringing the 672 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone 3G, and Apple iPhone 3GS, that infringe one or more claims of the 672 Patent.

124. Apple is inducing the infringement of the 672 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

125. Apple is contributing to the infringement to the infringement of the 672 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

126. Apple's infringement of the 672 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

127. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT X
INFRINGEMENT OF U.S. PATENT NO. 7,403,621

128. Nokia incorporates by reference the allegations set forth in Paragraphs 1-127 of this Complaint as though fully set forth herein.

129. Apple has infringed and is infringing the 621 Patent by making, using, offering for sale, and selling in the United States, without authority, products and services including wireless communication devices such as the Apple iPhone, the Apple iPhone 3G, and Apple iPhone 3GS, that are covered by one or more claims of the 621 Patent.

130. Apple is inducing the infringement of the 621 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

131. Apple is contributing to the infringement to the infringement of the 621 Patent by others in the United States. The direct infringement occurs by the activities of end users of the accused products.

132. Apple's infringement of the 621 Patent is exceptional and entitles Nokia to attorneys' fees and costs incurred in prosecuting this action under 35 U.S.C. § 285.

133. Apple's acts of infringement have caused damage to Nokia and Nokia is entitled to recover from Apple F/RAND compensation as a result of Apple's wrongful acts in an amount subject to proof at trial, and such other relief as may be appropriate.

COUNT XI
DECLARATORY JUDGMENT REGARDING F/RAND RIGHTS

134. Nokia incorporates by reference the allegations set forth in Paragraphs 1-133 of this Complaint as though fully set forth herein.

135. The patents-in-suit are infringed, not invalid, and enforceable.

136. Prior to filing this lawsuit, Nokia made various offers to Apple for a license to make, use, offer to sell, and/or sell products embodying the claims of the patents-in-suit, and/or the methods of the claims of the patents-in-suit.

137. Nokia has met its obligations under its F/RAND undertakings through, among other things, the offers made by Nokia to Apple.

138. Despite Nokia's offers for the F/RAND terms and conditions for a license under the patents-in-suit to Apple, Apple has refused to compensate Nokia on F/RAND terms for the use of the patents-in-suit in breach of its obligation to pay for the use of Nokia's patents.

139. Apple's continued use of the patents-in-suit without paying F/RAND compensation has caused and will continue to cause Nokia irreparable harm unless enjoined by the Court until Apple pays to Nokia F/RAND compensation for past infringement, and irrevocably commits to payment of such compensation in the future.

140. Once the appropriate compensation on F/RAND terms is determined, Apple should be enjoined from importing, making, using, selling, or offering for sale products and services embodying the claimed inventions of the patents-in-suit until and unless it pays

Nokia F/RAND compensation for past infringement, and irrevocably commits to payment of such compensation in the future.

141. This Court's equitable powers are hereby invoked by this Count, and Nokia accordingly requests that the Court consider such other relief, equitable or otherwise, as it may find appropriate at the time for entry of judgment in this case.

DEMAND FOR JURY TRIAL

142. Pursuant to Rule 38(b) of the Federal Rules of Civil Procedure, Nokia demands a trial by jury of this action.

PRAYER FOR RELIEF

WHEREFORE, Nokia prays for judgment and seeks relief against Apple as follows:

(a) For judgment that the patents-in-suit have been and continue to be directly and/or indirectly infringed by Apple;

(b) For judgment that the patents-in-suit are not invalid and are enforceable;

(c) For judgment that Nokia has complied with its legal obligations with respect to negotiating F/RAND terms and conditions of licenses to the patents-in-suit to Apple;

(d) For judgment that Apple has refused to compensate Nokia on a F/RAND basis for Apple's use of the patents-in-suit;

(e) Once the appropriate F/RAND compensation is determined, for a permanent injunction preventing further infringement, contributory infringement, and inducement of infringement until and unless Apple pays to Nokia such F/RAND compensation for past infringement, and irrevocably commits to payment of such compensation in the future.;

(f) For actual F/RAND damages together with prejudgment interest;

(g) For an award of attorneys' fees pursuant to 35 U.S.C. § 285 or as otherwise permitted by law;

(h) For all costs of suit; and

(i) For such other and further relief as the Court may deem just and proper.

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October 22, 2009

EXHIBIT A

US005802465A

United States Patent [19]

Hamalainen et al.

[11] **Patent Number: 5,802,465**

[45] **Date of Patent: Sep. 1, 1998**

[54] **DATA TRANSMISSION IN A RADIO TELEPHONE NETWORK**

[75] Inventors: **Jari Hamalainen**, Tampere; **Timo Jokiaho**, Vantaa, both of Finland

[73] Assignee: **Nokia Mobile Phones Ltd.**, United Kingdom

[21] Appl. No.: **724,375**

[22] Filed: **Oct. 1, 1996**

Related U.S. Application Data

[63] Continuation of Ser. No. 301,340, Sep. 6, 1994, abandoned.

[30] **Foreign Application Priority Data**

Sep. 6, 1993 [FI] Finland 933894

[51] **Int. Cl.**[6] **H04Q 7/20**

[52] **U.S. Cl.** **455/403**; 455/452; 455/560

[58] **Field of Search** 455/403, 422, 455/450, 452, 455, 509, 516, 550, 560, 435; 370/389, 338

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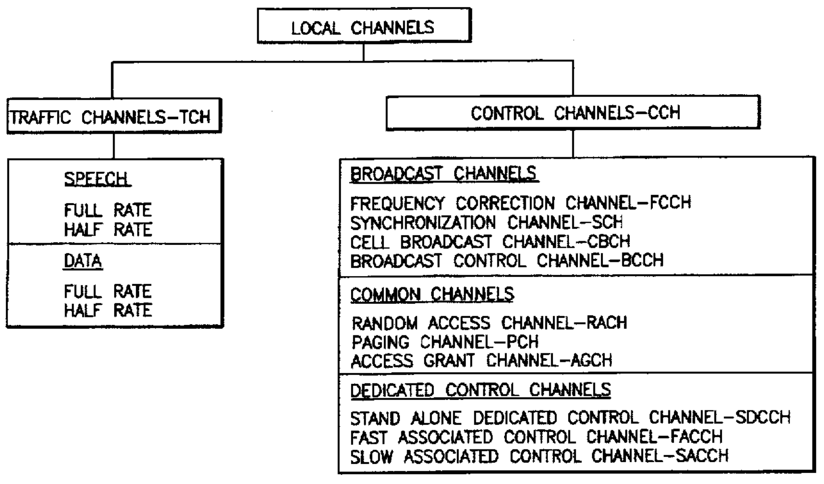
Primary Examiner—Dwayne D. Bost
Assistant Examiner—William G. Trost
Attorney, Agent, or Firm—Perman & Green, LLP

[57] **ABSTRACT**

For bidirectional transmission of packet data, a packet data service unit (Agent) is disposed in a digital cellular system connected to be in association with a Mobile Switching Center, and connecting the cellular network to the date network. As a mobile station is connected to the packet data service unit, signalling related to connection formation characteristics of the network is first accomplished. As a result thereof, the mobile station and the data service unit are provided with a number of stored parameters relating to each other. This situation creates or is called a virtual channel. When a mobile station wants to transmit or receive data packets between the mobile station and the data service unit a packet data transfer channel is established making use of the parameters of the virtual channel and thereby using substantially less signalling than the channel establishment signalling characteristic of the network, one part thereof being a radio channel and the other part a time slot in a digital trunk line. On termination of data packet transfer, at least said radio channel is disassembled but the virtual channel is maintained until the disconnection of the mobile station from the data service.

32 Claims, 7 Drawing Sheets



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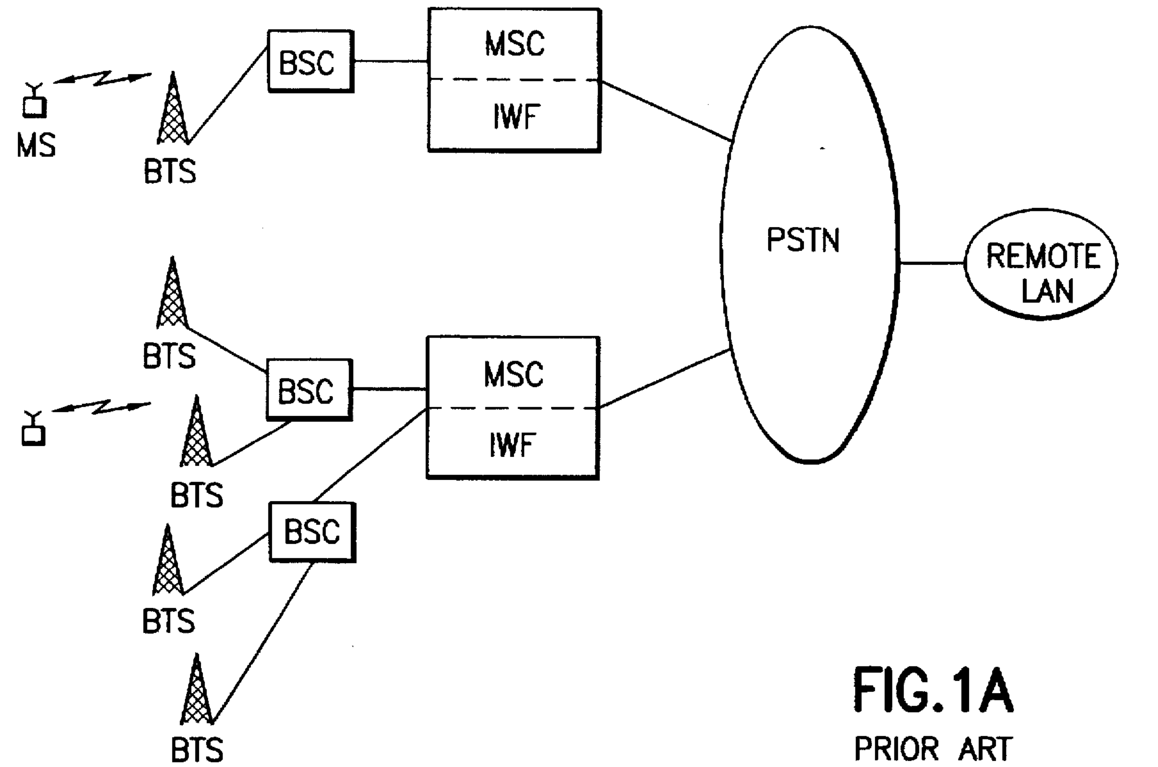

FIG. 1A
PRIOR ART

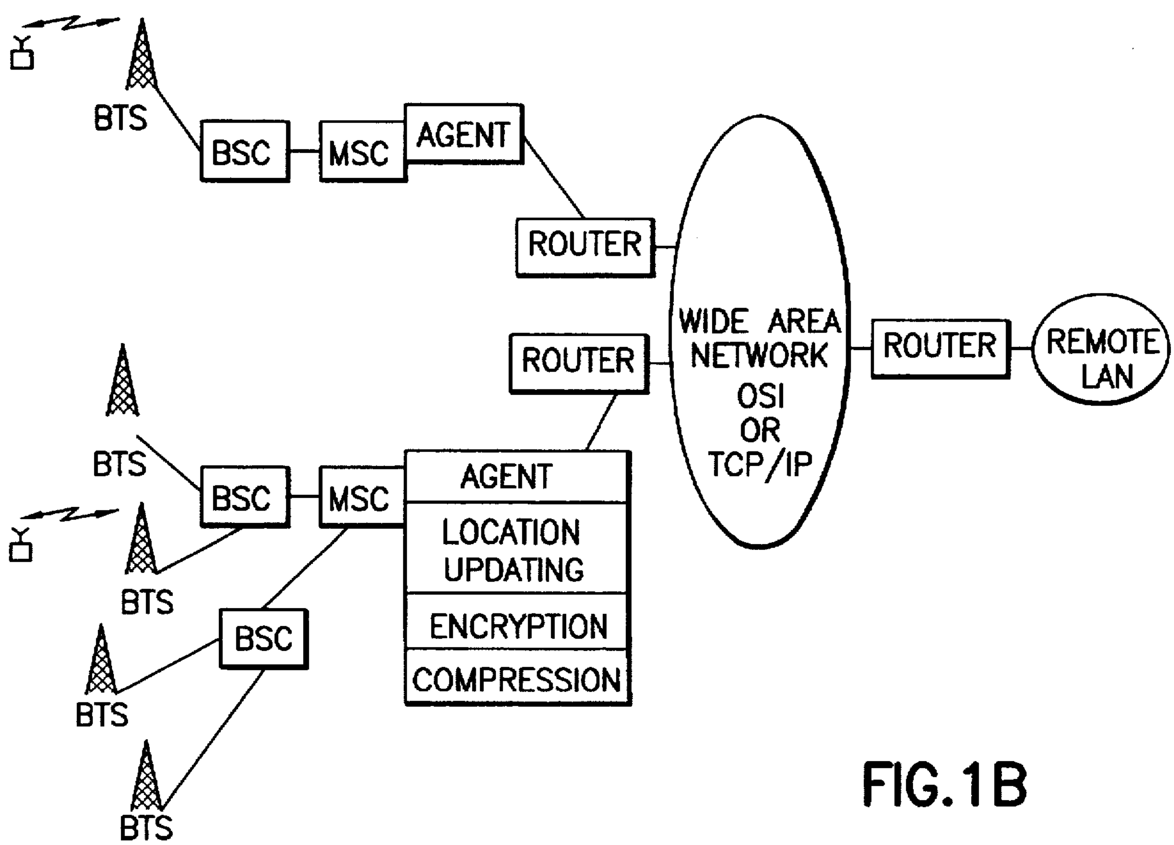

FIG. 1B

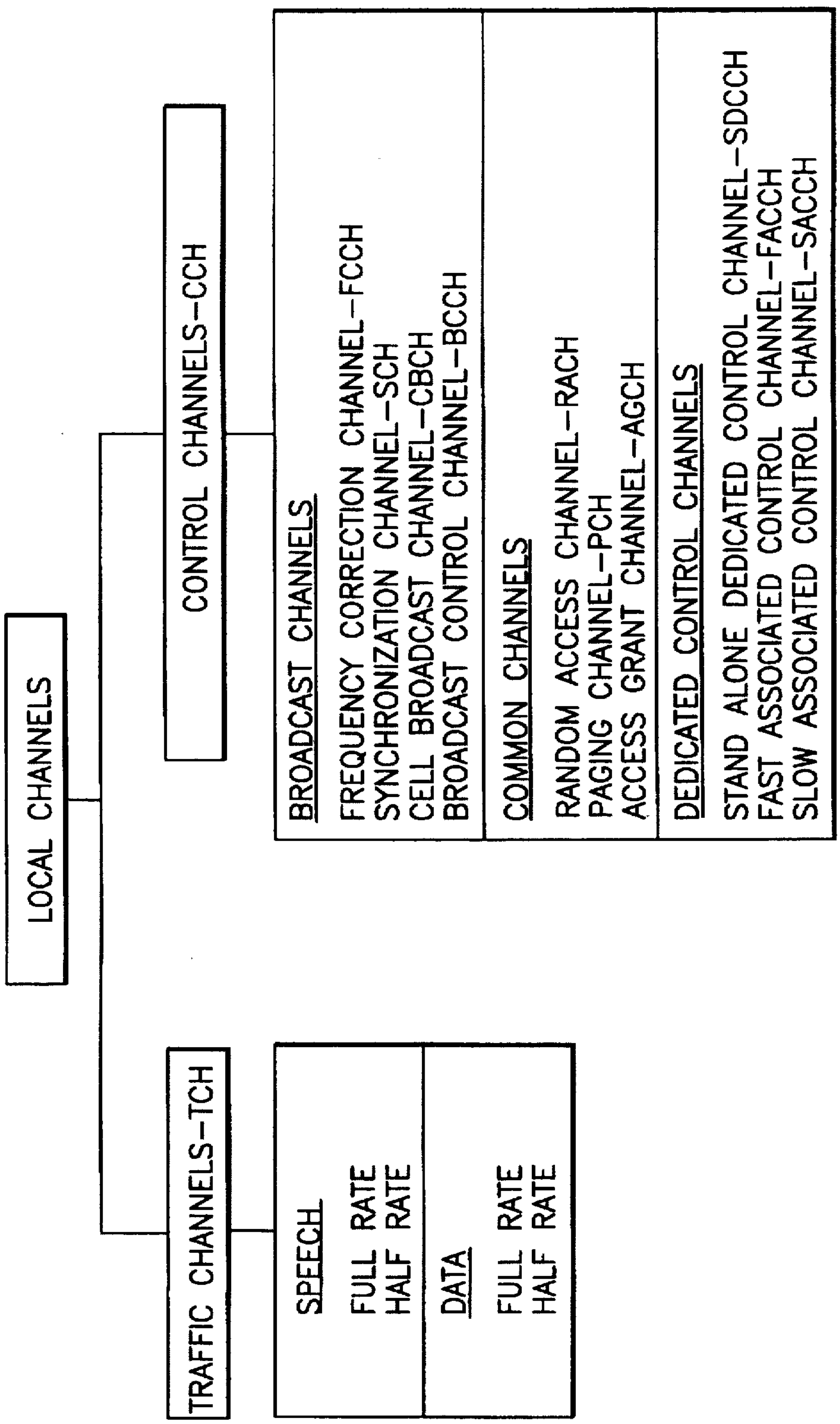
LOCAL CHANNELS
CONTROL CHANNELS–CCH
TRAFFIC CHANNELS–TCH
BROADCAST CHANNELS
FREQUENCY CORRECTION CHANNEL–FCCH
SYNCHRONIZATION CHANNEL–SCH
CELL BROADCAST CHANNEL–CBCH
BROADCAST CONTROL CHANNEL–BCCH
COMMON CHANNELS
RANDOM ACCESS CHANNEL–RACH
PAGING CHANNEL–PCH
ACCESS GRANT CHANNEL–AGCH
DEDICATED CONTROL CHANNELS
STAND ALONE DEDICATED CONTROL CHANNEL–SDCCH
FAST ASSOCIATED CONTROL CHANNEL–FACCH
SLOW ASSOCIATED CONTROL CHANNEL–SACCH
SPEECH
FULL RATE
HALF RATE
DATA
FULL RATE
HALF RATE

FIG.2


0 0 1
RANDOM REFERENCE

FIG.3

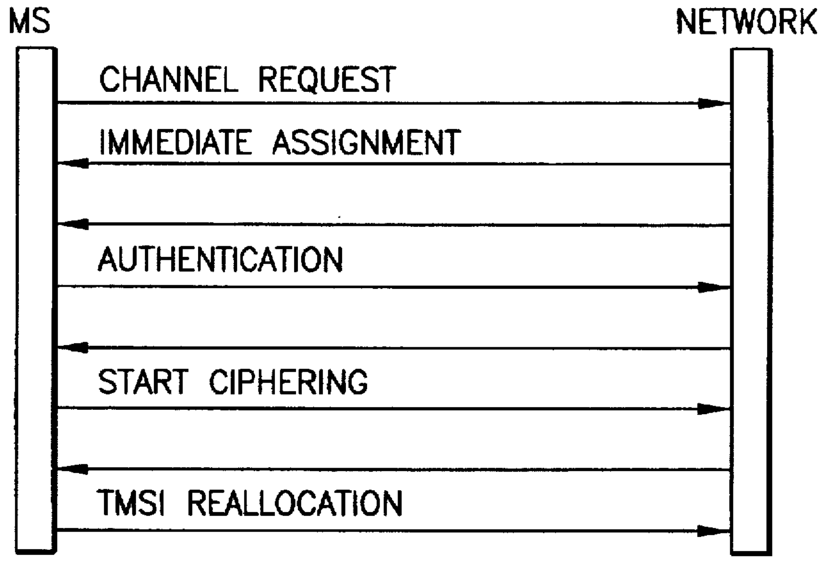

FIG.4

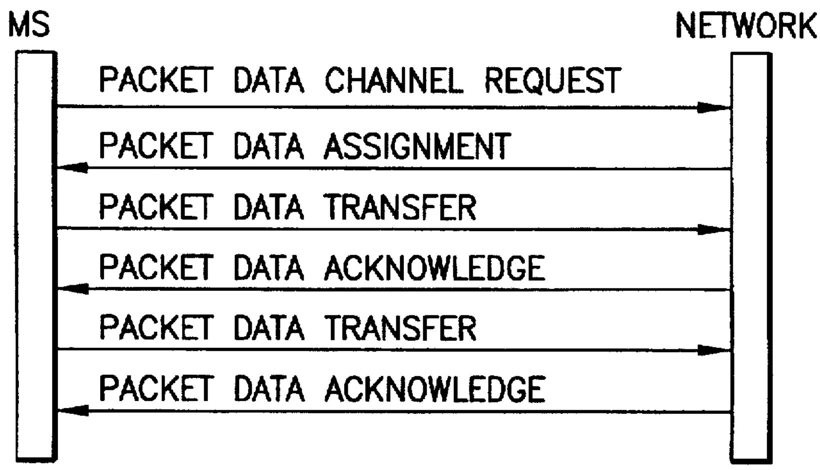

FIG.5

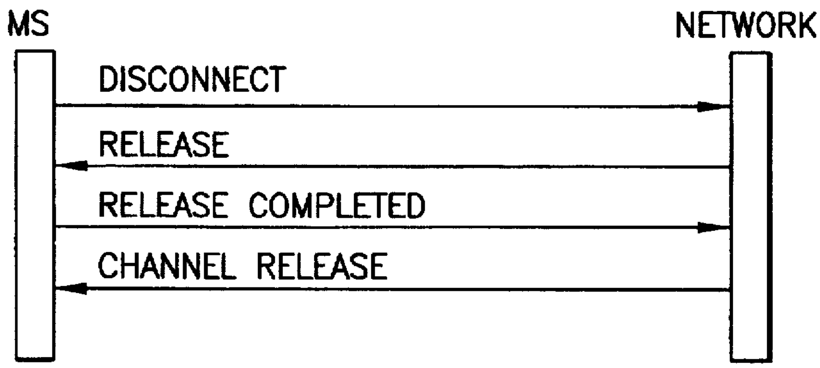

FIG.6

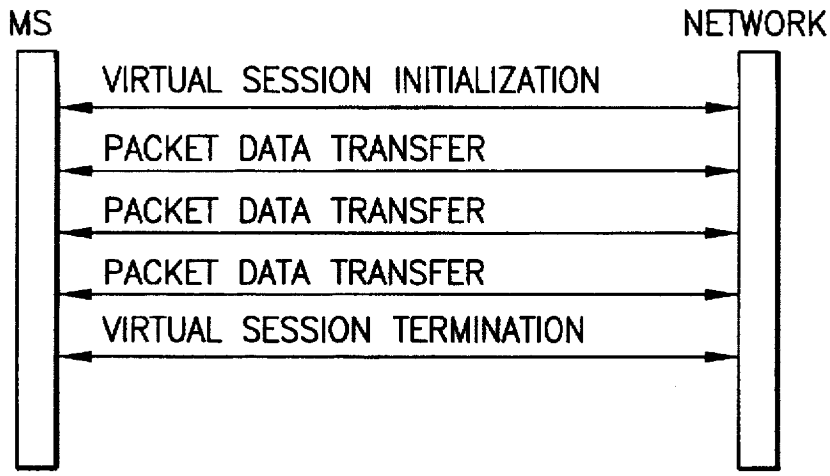

FIG.7

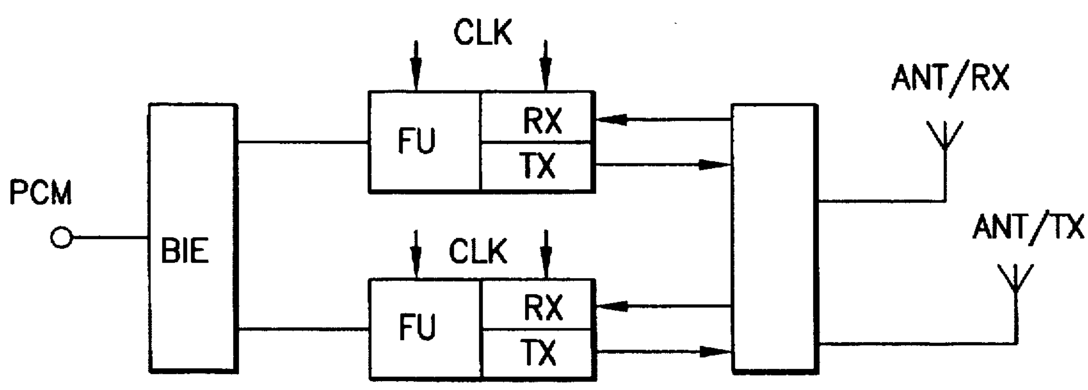

FIG.8

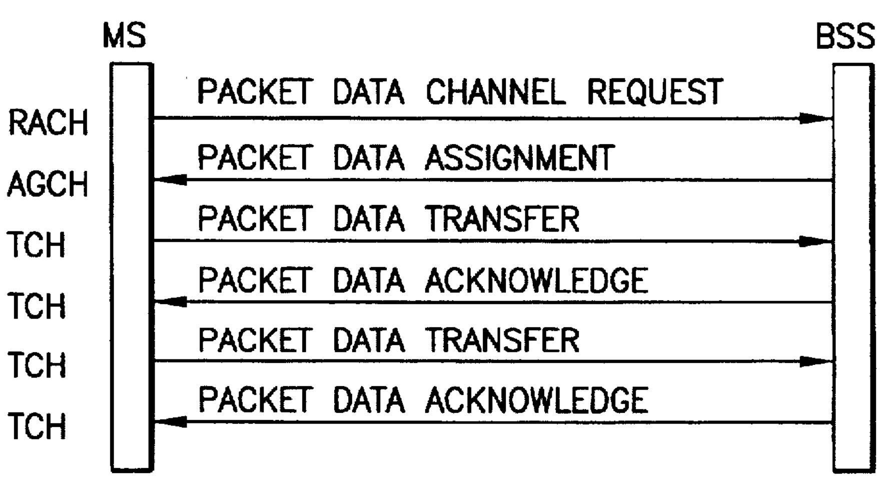

FIG.9

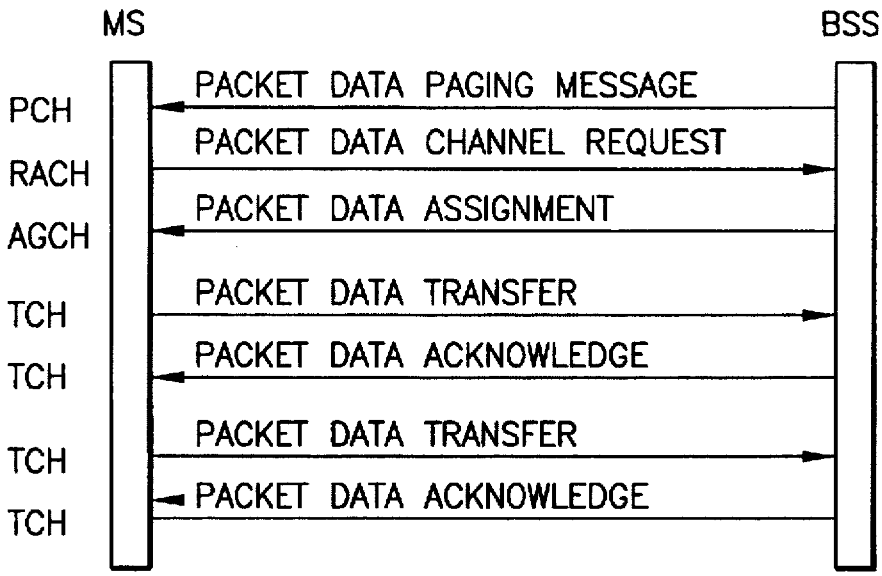

FIG.10

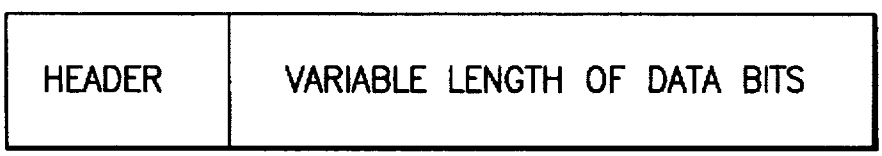

FIG.11

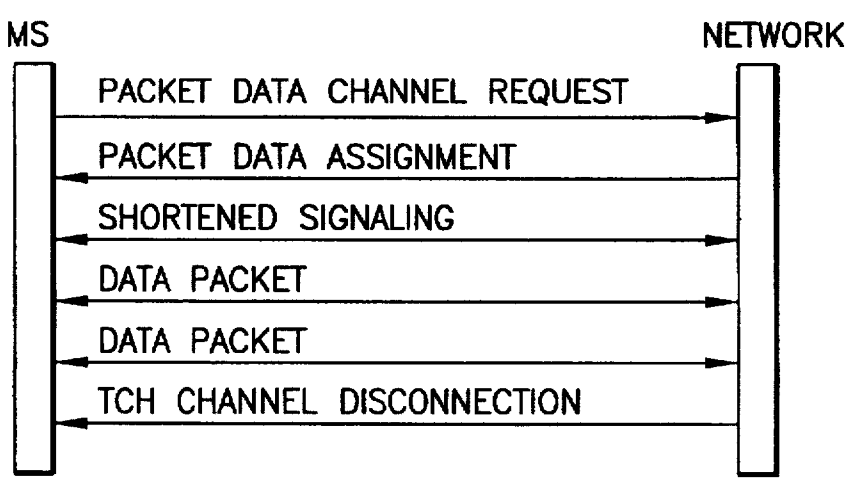

FIG.15

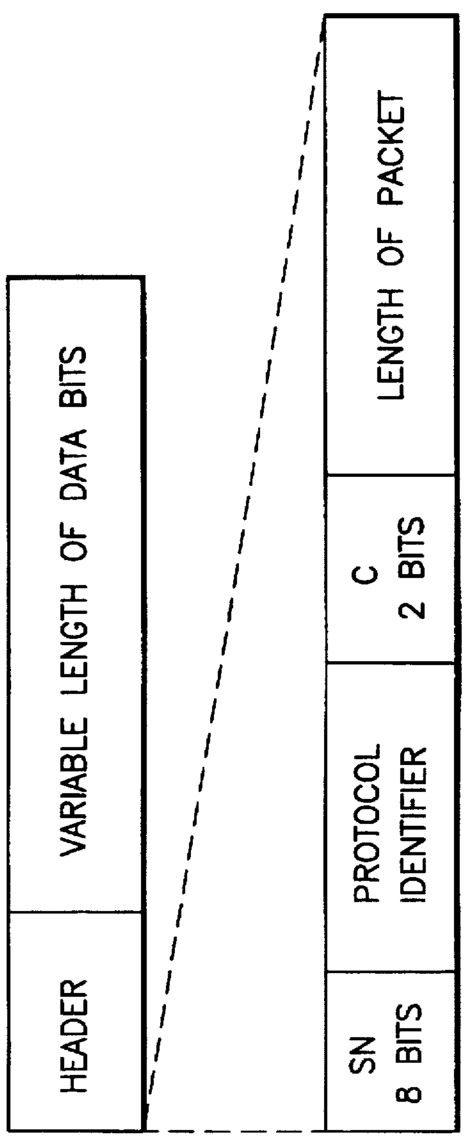
HEADER
VARIABLE LENGTH OF DATA BITS
SN
8 BITS
PROTOCOL
IDENTIFIER
C
2 BITS
LENGTH OF PACKET

FIG.12

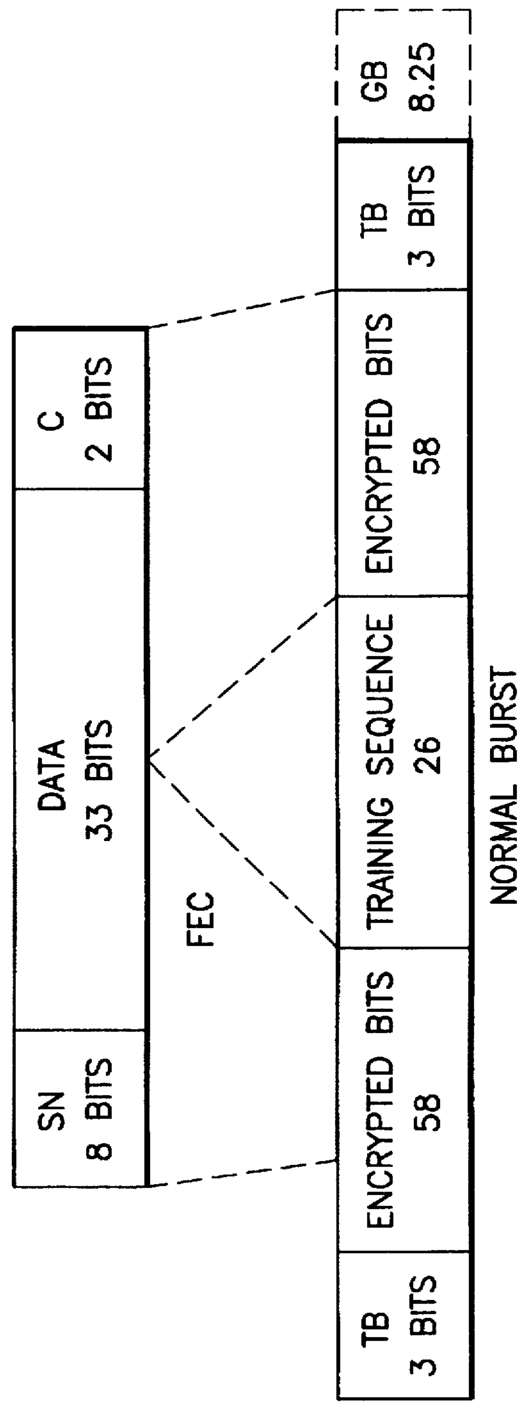
SN
8 BITS
DATA
33 BITS
C
2 BITS
FEC
TB
3 BITS
ENCRYPTED BITS
58
TRAINING SEQUENCE
26
ENCRYPTED BITS
58
TB
3 BITS
GB
8.25
NORMAL BURST

FIG.13

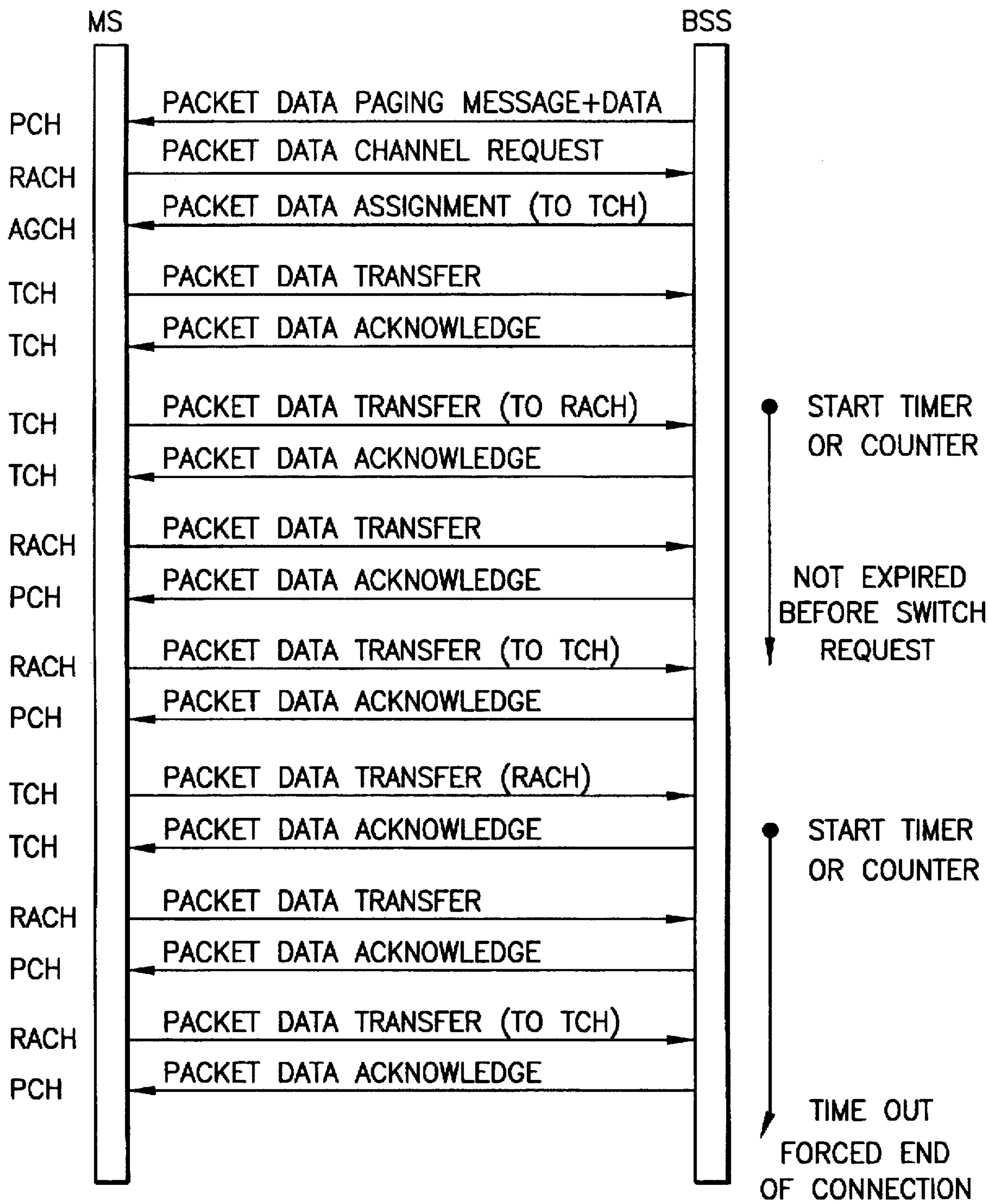
MS
BSS
PCH
PACKET DATA PAGING MESSAGE+DATA
RACH
PACKET DATA CHANNEL REQUEST
AGCH
PACKET DATA ASSIGNMENT (TO TCH)
TCH
PACKET DATA TRANSFER
TCH
PACKET DATA ACKNOWLEDGE
TCH
PACKET DATA TRANSFER (TO RACH)
START TIMER OR COUNTER
TCH
PACKET DATA ACKNOWLEDGE
RACH
PACKET DATA TRANSFER
PCH
PACKET DATA ACKNOWLEDGE
NOT EXPIRED BEFORE SWITCH REQUEST
RACH
PACKET DATA TRANSFER (TO TCH)
PCH
PACKET DATA ACKNOWLEDGE
TCH
PACKET DATA TRANSFER (RACH)
TCH
PACKET DATA ACKNOWLEDGE
START TIMER OR COUNTER
RACH
PACKET DATA TRANSFER
PCH
PACKET DATA ACKNOWLEDGE
RACH
PACKET DATA TRANSFER (TO TCH)
PCH
PACKET DATA ACKNOWLEDGE
TIME OUT
FORCED END OF CONNECTION

FIG.14

DATA TRANSMISSION IN A RADIO TELEPHONE NETWORK

This is a continuation of application Ser. No. 08/301,340 filed on Sep. 6, 1994, now abandoned.

FIELD OF INVENTION

The present invention relates to the transmission of data in a radio telephone network.

BACKGROUND OF THE INVENTION

An example of a radio telephone network, in this case a conventional digital cellular network, is shown in FIG. 1A. The network comprises Base Station Controllers (BSC), each of which control a number of Base Transceiver Stations (BTS). The BTS and mobile stations (MS) are connected via a radio communications channel. A Base Station Controller and the base stations with which it is connected form a Base Station Subsystem. The BSCs are connected to Mobile Switching Centres (MSC) via digital trunk lines which control the Base Station Subsystems. The MSCs route communication traffic to a general Public Service Telephone Network (PSTN) or private networks such as a Local Area Network (LAN). A Base Station Controller may also be physically located with the Mobile Switching Center. The service range of a base station forms a cell and a mobile station within the service range is typically served by that base station. The mobile station is able to move from one cell to another and roam from under the control of one Base Station Controller to be under the control of another Controller without losing a connection to the radio telephone network.

In known cellular networks data information can be transmitted between the home network of a mobile station and a terminal or destination network. The terminal network can include a home network, another network of the same system, a fixed telephone network, or a data network. The network services typically include synchronous and asynchronous circuit-switched data transfer from the cellular network to the external telephone network PSTN, to a circuit-switched date network or an ISDN network. Suggestions have also been made on implementing asynchronous packet switching to an external packet switched data network.

As shown in FIG. 1A, data transmitted by a mobile station enters a data Inter Working Functions unit, IWF, associated with the Mobile Switching Centre, from there via a modem to the Centre wherefrom it is further transmitted, e.g., via the PSTN, to a target means or target data network, such as a private LAN network. The transition network is thus the general telephone network.

A typical method of data transmission between networks and also within a network is circuit switching, in which a transfer channel is established for the transfer of data. Establishing a channel is a time-consuming operation and requires a lot of signalling, such as sending a control channel request and assignment of a channel, authentication checks, installation of an encrypting mode and others, before the channel is set up for transferring data information. Circuit switching, when applied for data transfer, is uneconomical since the transfer needs a wide frequency band. Also a user is charged irrespective of whether data is transmitted or not. This is because in a circuit-switched network the channel has to be maintained until all data information has been transmitted, which regarding the capacity is uneconomical. Since charging of the user is usually based on the length of

the reserved connection time in the circuit-switched network, the user is obliged to pay for "nil" because the time used for the actual data transfer is a minor part of the total connection time. Typically, cellular networks have primarily been optimized for speech transfer, and for that purpose, circuit-switched data transfer is appropriate.

In a digital cellular network, such as in the European GSM network and in the American network of the EIA/TIA (Electronic Industries Association/Telecommunication Industry Association) standards, suggestions have been made on data communication as packets, as so-called packet data e.g., in the patent U.S. Pat. No. 4,887,265. This patent discloses a system in which several mobile stations send packet data to one base station using the same channel. When the Base Station Controller receives an assignment request for a data channel from the mobile station, it transmits a channel assignment to the mobile station, whereby the mobile station moves on that data channel. The same channel is also available for use for all other mobile stations within the range of said cell. A request, a channel assignment and transfer on a channel require a considerable amount of signalling. Handover of a data connection from one base station to another is also possible in this system. In the system disclosed by the patent, a permanent channel is provided for packet transfer, being constantly available, irrespective of a momentary need.

SUMMARY OF THE INVENTION

According to a first aspect of the invention there is provided a radio telephone system comprising:

a mobile station; and

a fixed station, wherein a parameter of the mobile station for setting up a data communication channel is capable of being stored by the fixed station and a parameter of the fixed station for setting up a data communication channel is capable of being stored by the mobile station, for forming a virtual data communication channel between the mobile station and the fixed station, thereby expediting establishment of a real data communication channel.

According to a second aspect of the invention there is provided a method of transmitting data in a radio telephone network comprising:

storing a parameter of a mobile station for setting up a date communication channel at a fixed station; and

storing a parameter of the fixed station for setting up a data communication channel at the mobile station, for forming a virtual data communication channel between the mobile station and the fixed station, thereby expediting establishment of a real data communication channel.

According to a third aspect of the invention there is provided a radio telephone adapted to store a parameter for setting up a communication channel of a fixed station for forming a virtual data communication channel with the fixed station thereby expediting establishment of a real data communication channel.

These aspects of the invention provide the advantage that a real data communication channel can be established quickly and when a mobile station desires to transmit data. In between the transmission of data the real data communication channel can be switched to a virtual data communication channel ready for quick reestablishment. Thus, a communication channel does not have to be continually open, even during no actual transmission of data. Thus, the costs of transmitting data are reduced.

Alternatively, the virtual data communication channel can be formed if a mobile station having data transmission capability registers with the fixed station, or if a mobile station registered with the fixed station requests a data communication channel. An advantage of forming a virtual data communication channel only when a mobile station requests a data communication channel is that unnecessary signalling is avoided.

Optionally, the data communication channel can be a channel usually reserved for speech transmissions, or signalling or control transmissions. A particular advantage of using signalling or control channels is that the transmission of data does not reduce the number of speech channels available to the users of the system.

Advantageously, the data communication channel is adapted for transmitting packet data, which is a transmission form particularly suitable for use with a data communication channel which can quickly be opened or closed.

Another advantage is that data packets can be created at the mobile stations and transferred directly to a data network without the need for transition networks, such as Packet Assembler/Disassemblers (PADs) or using the PSTN. Additionally, the mobile station itself can receive packet data, i.e. the system is bidirectional.

An appropriate existing cellular system currently in use is, for instance, the European GSM system.

Embodiments of the invention will now be described by way of example only and with reference to the drawings, in which:

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1A presents a cellular network in accordance with the prior art,

FIG. 1B presents a cellular network in accordance with the present invention

FIG. 2 is a schematic presentation of the logical channels of the GSM system,

FIG. 3 illustrates the configuration of a channel request,

FIG. 4 presents the starting signalling of a virtual channel,

FIG. 5 presents the steps of transferring packet data,

FIG. 6 presents the terminating signalling of the virtual channel,

FIG. 7 presents a phase after a channel has been assembled,

FIG. 8 is a diagrammatic presentation of a base station,

FIG. 9 illustrates mobile phone originated data transfer,

FIG. 10 illustrates data transfer terminating in a mobile phone,

FIG. 11 presents a format of a packet data message,

FIG. 12 presents a format of another packet data message,

FIG. 13 presents an order of a RACH frame for a standard burst,

FIG. 14 presents the phases of a packet data transfer, and

FIG. 15 presents signalling when a connection is broken at interfaces.

DETAILED DESCRIPTION OF EMBODIMENTS OF THE INVENTION

In a particular example of a cellular network, the physical channel of a mobile station and a base station, that is, a radio frequency channel, consists of consecutive frames which in turn consist of time slots, in one of which the transmission is performed, the reception in another, in another listening to paging calls, etc. The respective time slots constitute a logical channel, of which a great number may be available.

In digital cellular networks, a mobile station can send and receive data on a traffic channel particularly intended for speech and data transfer. Both of them cannot be transmitted simultaneously but the user or the network makes a selection which thereof is to be transferred. Data as well as speech are sent as bursts on a radio channel. This means that in a transmission time slot a brief data burst is transmitted in the middle of the time slot so that a considerable part of the total time used for transmission means the time between the bursts when no information is transferred.

A particular type of data service known as packet data service has been defined in the GSM network. In this service, the number selected by a data transmitting mobile station informs the network that a circuit-switched connection has to be created to a packet assembling or disassembling unit performing the connection with a data network, such as X.25, which can be a Packet Assembler/Disassembler (PAD) or a Packet Handler (PH). The Packet Assembler/Disassembler can be placed in association with or also behind the ISDN network. The mobile station sends data as continuous data flow, not as packets, to PAD or PH, which forms the data packets and transmits them onwards via the data network to the target. If PH is a so-called Basic Packet Handler, the data connection is always located via a given point PH, even in any network. The Basic Data Handler also supports the mobile terminated direction in data transmission. On the other hand, the mobile terminated direction is not supported by the so-called dedicated Packet Handler, nor PAD. The traffic between the mobile station and the packaging means imitates synchronous or asynchronous data transfer, wherebelow a radio traffic protocol RLP is located.

In the data packet service of the GSM network no packets are produced in the mobile stations, but in PAD. The traffic is unidirectional also in the sense that the connection is mobile station-originated, i.e., the station should send a request to the network for creation of a data connection. No packets can be sent to the mobile station unless the station itself has first requested the opening of a line. It is also to be noted that data is conducted via the telephone network, the pricing of the data transfer whereof being much higher than pricing of transfer within a data network.

The sending and reception function of data packets can be arranged to be positioned in all mobile stations or in some of them only. For the mobile stations without such function, a packet data transfer is to be completely opaque so that mobile stations of different types are enabled to function without any problems in the network simultaneously. Thus, the packet data feature is an additional service provided by the network, though requiring that the mobile station possesses a property to use such service. The implementation of the system must be such that it requires only a few changes in digital cellular systems in current use and, as an additional feature, it is well appropriate for use in current systems such as GSM, DCS 1800 operating in 1.8 GHz range or PCN.

In new networks a so-called Short Message Service is most often determined, wherewith a mobile station is enabled to transmit and receive temporally short messages. A transfer of a short message requires, however, standard connection formation routines, thus requiring part of the frequency band and limiting the amount of data to be transferred.

For transferring packet data no allocated radio channel and data route via the network are maintained continuously.

In accordance with the invention, a virtual channel is arranged for data packet transfer in the network between the mobile station and the Mobile Switching Center. When a phone provided with a packet data function enters the range of the Mobile Switching Center, assigned as a user of the packet data service, all necessary signalling is executed, whereafter the Center, or more specifically, the packet data service unit (Agent) in association therewith is provided with all the information it needs concerning the phone and establishing a true transfer channel. Such data, containing in fact information about the location of the phone, is called a virtual channel. The virtual channel is thus a virtual connection between the mobile station and the data service unit, enabling fast transition into data transfer mode, paying regard to the parameters stored in the memories of the mobile station and the data service unit. When packet data has to be transferred from the phone to the network, or vice versa, no complete signalling is needed between the phone and the Center, since that was carried out earlier; instead, a true transfer route can be set up extremely fast and with very low-level signalling between the mobile station and the packet data service unit (Agent), whereupon the packets are transferred. The transfer route, or at least the radio channel, is released as soon as there is no packet data to be transferred. Instead, the virtual channel is kept in constant preparedness as long as the mobile station is listed in the data service. In accordance with the present invention, a very rapid connection to the packet data transfer mode can be made, and the transfer route is kept reserved only when there is something to be transferred.

A means to control the transfer of packet data is arranged to be in conjunction with the Mobile Switching Center, and is known as a data service unit (Agent), which can be a computer or a process. It is a data service center provided with a number of connection services and which has access to other networks and the services thereof. The Agent has been placed logically in association with a Mobile Switching Center MSC, though the physical location can be inside the Center as part of the processes thereof or outside the Center in the form of one or more computers connected via a transmission link to the Center. The basis of the Agent is an Interface Unit IFU connecting the cellular network to another network, such as to TCP/IP or OSI networks (TCP=Transmission Control Protocol, IP=Internet Protocol, OSI=Open Systems Interconnection). Thus, a mobile station MS provided with a packet data function communicates by means of the data service unit (Agent) with the other networks, and the virtual channel is placed specifically between it and the data service unit (Agent). Therefore, each mobile station utilizing the packet data service under the control of the Mobile Switching Center is supervised by the data service unit (Agent) in association with the Mobile Switching Center.

The Agent performs at least some of the following functions: It

- registers all telephones provided with a packet data function under the control of the Mobile Switching Center,
- informs the phone of a message to arrive,
- removes the phone from the register after terminating of connection,
- transfers the messages of the phone to the rest of the network,
- transfers the messages from the rest of the network to the phone,
- buffers messages with a view to efficient transmission via the network,

- when necessary, performs encrypting/decrypting,
- when necessary, performs compression/decompression between the phone and the Agent,
- updates the data base thereof (location updating),
- receives messages addressed to the paging channel.

Normally, the virtual channel is initialized when the user starts using the packet data service, and the channel is terminated after the user leaves the service. During the time between the start and the termination, i.e., while being connected with the service, the mobile station is able to move and transfer from one cell to another. The handover function prerequires disassembling the virtual channel and assembling a new one. The handover is practically unobservable by the user.

When entering the cell area, a mobile station listens to the System Info channel, characteristic of all cellular networks and constant transmission from the base station, being therethrough informed if the packet data service is in use in the network or in the cell. A System Info message may include an identification referring e.g., to the packet data service. When a mobile station wants to be connected to a packet data service, it transmits via a base station to the network a request for setting up a virtual channel. The request commences in the network a standard control signalling sequence utilized thereby and characteristic of said network, in which the authentication of the requester is checked, encrypting is started and the requester is provided with an interim identification number. The Agent in association with the Mobile Switching Center controlling the packet service, is also informed, whereby it includes the supervision of the mobile phone under the control thereof. The Mobile Switching Center maintains an ongoing register on the location of the mobile station, whereby handover from one cell to another is possible and a fast preparedness to transfer to data transmission or reception exists because there is no need for the phone to request separately for a traffic channel.

Instead of a System Info message, it is also possible to operate so that the mobile station requests the network via a short message service whether the packet data function is engaged. The network responds by an equal message of the short message service. The short message services (SMS) are a service mostly included in the digital networks.

The control signalling associated with the management of the data connection between the data service unit (Agent) in association with the Mobile Station and the Mobile Switching Center MSC is executed along with the data messages in the signalling plane. The functions in the signalling plane are provided with functions for setting up, maintaining and terminating a connection between the cellular network and the other networks. It also includes functions for updating the register, authentication, and a function for providing an interim subscriber number TMSI.

A plurality of protocols are available for use in the transfer of data packets between the mobile station and the data service unit (Agent). The radio interface sets, however, certain limits, such as a requirement for minimizing the amount of data transmitted across the interface. The amount can be minimized by compressing the data section of the packets. The data are compressed prior to transmission, e.g., by means of V.42bis compression algorithm, and the receiver decompresses the data using the same algorithm. Also the bit amount in the header of the data packets may be reduced. Such functions are attended to by a Virtual Channel Protocol, which also attends to the control messages between the agent and the mobile station, as well as adapts the packets of the upper protocols into the Radio Link

Protocol (RLP) frames. A paging message transmitted to all mobile stations (broadcast) of the cell or to certain mobile stations (multicast) is transmitted on the data section of the broadcast or the multicast protocols, respectively.

After the virtual channel has been assembled between the mobile station and the base station, the mobile station can neither start nor receive ordinary calls. Instead, the transmission and reception of short messages SMS is possible.

When wishing to transmit data packets, a mobile stations ends a request to the network for channel assignment. Since the majority of the signalling needed in establishing a channel has been already executed at the beginning of creating the virtual connection, the setting up of a data packet transfer channel extending from the mobile station to the Agent, required at this moment, is fast. This means a short time from the channel assignment request to transmission of packets.

The transmission may be accomplished according to a first or second embodiment of the invention. When a user of the mobile station switches off the packet data function on termination of data transmission or when the network terminates the connection, the data route is disassembled and the radio channel is released; optionally, the virtual channel may be maintained.

A packet data session refers to the time commencing when a user starts a packet data function (informs of his desire to be connected to the service), and ending when the user terminates the service. In the course of the session the user may transmit packets both to a terminal network and receive them from the source network. Roaming and handover are possible. In the course of a session one or several virtual channel connections are created, though only one at a time.

In accordance with a first embodiment, the radio channel for a data route is a standard traffic channel of a cellular system which is intended for transfer of speech and non-packet shape data via broadcasting between a mobile station and a base station. When wishing to transmit data from a mobile station (i.e., mobile originated), the station requests the network via a base station for a channel using the same signalling channel as normally used when the station sends a request to connect a call. The signalling channel is a random access channel which all mobile stations of the call use. The channel runs from the mobile station to the base station, that is, it is a so-called uplink direction channel. Due to the random access, collisions may occur when channel requests enter simultaneously. In such an event the request has to be repeated. The request message includes a special bit configuration, an identification block with which the station reports of a service it wants to have, such as speech, data, packet data; in the present case, the identification configuration indicates that the desired service is transmission of packet data.

After the network has processed the request and allocated the traffic channel, it transmits to the mobile station on the signalling channel a response containing information as to which traffic channel the station should move onto in order to transmit packet data. The channel on which the network responds to channel requests is a common Access Grant CHannel and is in a downlink direction. The mobile station tunes its transmitter onto the allocated traffic channel, and immediately starts transmitting packet data. The transmission lasts until all the data has been transmitted. The network may also start a particular counter or timer when the traffic channel has been allocated, whereby the transmission continues until the counter or timer expires. It is preferred to store the data to be transmitted in a buffer memory of the mobile station and to erase the memory by transmission.

When packet data is transferred according to the first embodiment via the network to a mobile station (mobile terminated transfer), the only difference from a transfer in the opposite direction is that the network informs the mobile station of a packet data transmission to come. For transmitting such information, a common paging channel is used. All mobile stations within the range of the cell continuously listen to this common downlink paging channel (speech pagings are transmitted on this channel). When the mobile station has received a message indicating that packet data is coming in, it acts in the same way as in the mobile-originated case: It transmits a traffic channel request to the base station, receives data on the channel, and moves immediately on to the traffic channel assigned thereto, thus being prepared to receive data packets. On termination of data flow, the network disassembles the traffic channel, so that it is released for use of other mobile stations present within the range of the cell. The data to be transmitted is preferably stored in a data buffer of the data service unit (Agent) and the buffer is erased all at once.

In accordance with the first embodiment, when transmitting packet data one traffic channel is reserved for such date which is normally used for transferring speech. On termination of transmission, the traffic channel is again free for use by any mobile station. The same mobile station may send another request for packet data transmission, whereby the sequence "channel request-transmission—channel release" can be repeated until the mobile station leaves the packet data service, and the virtual channel is disassembled.

In accordance with a second embodiment of the invention, a signalling channel or a control channel is used either exclusively or as an alternative to the use of the traffic channel for the transmission of packet data.

In accordance with the second embodiment, when a mobile station wishes to transmit data packets, i.e., mobile originated transfer, it sends a channel request page to a base station using the same random access channel upon which ordinary channel requests are transmitted. Said channel is in an uplink direction. All mobile stations of the cell employ the same channel for speech channel requests. The Mobile Switching Center decides, after receiving the request, which channel the mobile station should move to for data transmission. The channel can be either a standard traffic channel or a control channel. The control channel can be the same random access channel on which the channel requests are transferred from the mobile stations to the base station. The network establishes a traffic channel provided it has been selected to be the transfer channel. The base station transfers information to the mobile station on whether it is expected to use the standard traffic channel or the control channel for data transmission. Such information is transmitted on the Common Control CHannel, on the Access Grant CHannel, upon which channel the channel assignment is sent to the mobile stations. The mobile station moves to the traffic or control channel thus assigned, starting immediately to transmit packet data. In the course of the transmission, the channel may be handed over from the traffic channel to the control channel, and vice versa, even several times. On termination of transmission, the channel is disassembled and it is released for other uses. The transfer ends after a given time elapses or when a "packets over" message is received from the station.

If the network is required to transfer packet data to a mobile station, i.e., mobile terminated transfer, it informs the station via the standard common paging channel of a data packet transmission on the way. The paging includes a particular identification part (bit configuration) indicating that a packet data transfer is in question. In such paging the identification of a second mobile station has been replaced by the user's data section, including a packet coming in to the user from outside. If a packet from outside cannot be accommodated in one data section of the paging message, it is divided into several paging messages, all of which the mobile station receives, gathering one packet therefrom. When the mobile station has received the packet, it acts thereafter in the same way as when desiring to transmit data packets: it transmits a channel request to the base station, receives a channel assignment, moves on the assigned channel, the traffic channel or the control channel, and acknowledges the packet it has received.

The data route connection between the base station and the Agent connected with the Mobile Switching Center can be implemented in a number of ways. One possibility is to reserve a direct connection and to maintain the connection reserved continuously for packet data traffic. This means an ongoing existence of the connection so that no extra delays are formed. The connection can be a PCM time slot or several PCM time slots in the digital trunk line between the Base Station System (BSS) and the Mobile Station Center (MSC). When a mobile station provided with a packet data reception and transmission property enters the range of the cell in association with the base station, e.g., a BTS in FIG. 1A, the network immediately establishes a direct connection between the base station and the Mobile Switching Center provided for transmission of packet data. The connection can be one or several time slots in the PCM trunk line commonly used by all mobile stations provided with the packet data function. The entry of the mobile station into the cell is known because it has been transferred either as a result of a handover function, or, if entry from outside into the reception area is in question, or the phone is switched on, the phone is registered in the network.

In an embodiment such as the one described above the PCM channel within the network is constantly maintained but the radio route channel is reserved only when needed.

The use of the PCM time slots may also be optimized in that a direct connection is maintained only if the Base Station System (BSS) includes existing virtual connections, that is, at least one cell under the control of the Base Station Controller includes a mobile station connected to the packet data service, being in readiness to receive and transmit packet data. The direct connection is disconnected when no users of the service are found to be in the range of the BSS, and it is set up again when a first mobile station joins the packet data service.

A second possibility is that connections between the network and radio path are assembled and disassembled when need be. The examples described below include the connections provided according to the second possibility.

FIG. 1B shows a typical cellular network such as a GSM network provided with a data packet service in accordance with the invention. A data service unit (Agent) has been connected to a Mobile Switching Center, from where the packet data are conducted directly to a data network according to the OSI or TCP/IP protocol, and from there to a target network, such as a LAN. A difference between this network and the network of FIG. 1A lies in the fact that no data passes via the circuit-switched telephone network PSTN.

According to FIG. 2, the logical channels are divided into traffic channels TCH and control channels CCH. The traffic channels are intended for transferring coded speech and data. Each of them can be transferred at full rate or half rate. The control channels CCH are intended to transfer signalling and synchronization data, and three types of channels can be

distinguished thus: Broadcast Channels, Common Channels and Dedicated Control Channels. Below, "uplink" refers to the direction from a mobile station to a base station and "downlink" the direction from a base station to a mobile station.

The Broadcast Channels comprise the following:

- a Frequency Correction CHannel, FCCH, transferring frequency correction data to the mobile station, downlink,
- a Synchronization CHannel, SCH, transferring synchronization data to the mobile station and identification data of the base station, downlink
- a Cellular Broadcast CHannel, CBCH, short message service, bi-directional channel, and
- a Broadcast Control CHannel, BCCH, transferring general information on the base station, downlink.

The Common Channels comprise the following:

- a Random Access CHannel, RACH, uplink direction only, on which the mobile stations send a request for a dedicated channel
- a common Paging CHannel, PCH, whereby a base station sends a paging to a mobile station to inform of an incoming call, the channel being in downlink direction only,
- an Access Grant CHannel, AGCH, whereby the base station reports of a Stand-alone Dedicated Control CHannel, SDCCH, or directly of a Traffic CHannel, TCH, said channel being only downlink.

The Dedicated Control CHannels comprise the following:

- a Stand-alone Dedicated Control Channel, bi-directional, and
- a Slow Associated Control Channel and a Fast Associated Control Channel, the channels being bi-directional.

In accordance with the present invention, a Traffic CHannel (bidirectional), TCH, a Paging CHannel, PCH, (unidirectional, downlink), a Random Access CHannel, RACH, (unidirectional, uplink), and an Access Grant CHannel, AGCH, (unidirectional, downlink) are made use of. Channels of equivalent types can also be found in digital cellular systems other than GSM.

The mobile station listens to the Broadcast Transmission Control CHannel BCCH transmitted continuously by the base station of the cell and is therethrough informed of a packet data service being engaged in the network. Another procedure is that the mobile station requests on the Cellular Broadcast CHannel by transmitting a short message service whether the packet data function is in use in the network or not. The base station sends a short message response on the same channel.

When a mobile station sends a request to be a user of a packet data service, a message sequence as shown in FIG. 4 is carried out therebetween and the Mobile Switching Center. The events are read from top to the bottom. After the channel request is transmitted by the Mobile Station an immediate assignment of the control channel follows (FACCH), and on the assigned channel the authentication of the requester is checked (the network inquires on the authentication data and the mobile station sends a response), encryption is started, and an interim identification number TMSI is allocated. A Radio Link Protocol is established and maintained thereafter permanently. This means that in the course of a session the transmission of data packets can be performed without reassembling the radio link protocol. The data service unit (Agent) in association with the Mobile Switching Center controlling the packet data service is informed, thus transferring the control of the mobile station under the control of its own. The data service unit is now able to detect the mobile station and carry out the encryption and authentication without extra signalling. The virtual channel from the Mobile Switching Center to the mobile station has now been assembled. The radio link protocol is not disassembled before the end of the session (the phone is released from the data packet service) whereby the virtual channel is disassembled.

When a mobile station wants to transmit data, it transmits a request to set up a transfer channel for real packet data. The request is transmitted on a common Random Access CHannel RACH which is similar in configuration to the one shown in FIG. 3. By means of the first three bits of the message the nature of the connection is determined, and sequence 001 refers to a request to set up a data packet connection. The end of the message is a random reference number. The message is a modification of a standard GSM message. The base station receives the request, and after coding the sequence, it informs the mobile station on which control channel the signalling to be performed next is carried out and on which transfer channel the transfer of the packets is to take place. These phases are described by the two topmost phases in FIG. 15. The transmission channel has been assembled from the mobile station to the Base Station Controller. On a channel produced as above, the mobile station transmits first control messages, the third phase in FIG. 15, wherewith a data connection from the station to the data service unit (Agent) is provided, whereafter the channel from the mobile station to the Agent is complete for data transfer.

When a true channel, the first part thereof comprising a radio channel and the latter part a PCM time slot, has in the above described manner been established between the mobile station and the base station, the mobile station is able to transmit immediately packet data on that channel. After a demand on data transmission by the network the station transmits data packets, the network acknowledges the packets and sands requests for a repeated transmission if a transmission has been defective. The phases up to that point are presented in FIG. 5.

After transmission of all packets, the mobile station sends a request to the network to disassemble the true connection. After receiving the request the network sends an order to the mobile station to terminate the data activities, and the station acknowledges termination of those activities. The phases are presented in FIG. 6. The data packet transfer channel to the base station controller BSC and from there on to the data service unit (Agent) is disassembled. If a method based on direct PCM connection is used, the channel is left on.

Transfer of packet data may also be directed at the mobile station (i.e., it is mobile terminated). A base station sends on a common paging channel a paging to a mobile station, informing of a packet data transmission on the way. The mobile station then sends a channel request signal to the base station on the common Random Access CHannel RACH, whereby the process from that moment onwards is the same as in the mobile oriented case described above: establishing a virtual channel and immediate reception of packet data. In FIG. 7, each of the cases are presented step by step.

FIG. 8 shows a block diagram of a base station related to the present invention. The base station includes several parallel branches formed by the Framing Unit (FU) and the Transmitter/Receiver Unit RX/TX. A Base Band Interconnection Element (BIE) connects the base station to a digital PCM link. Part of the channels of the link are reserved for signalling and the rest for data transfer. The digital signals from the PCM link are conducted to the Framing Unit in

which they are arranged into TDMA frames, channel-coded, interlaced and transmitted as bursts onto the radio path via antenna TX. Prior to the transmission, the bursts have been modulated in the transceiver unit RX/TX and transferred to a carrier wave frequency. When the base station receives a TDMA signal from the mobile station, the signal is conducted via the necessary filters to the transceiver unit RX/TX where it is demodulated, transferred to a carrier frequency, and the modulation is indicated. The channel decoding and discharge of interlacing are performed in the Framing Unit (FU). Finally, the data signal is conducted to the PCM line and therefrom via the Mobile Switching Center to the receiving network.

The Base Station Controller produces all messages transmitted to the radio path, and all received messages are transferred via the Base Station to the base station Controller. Therefore, compared with the GSM currently used, the embodiment of the invention requires only minor changes in the software of the Base Station Controller. Changes have to be made also in the softwares of the mobile station and of the Center. The mobile station has to be able to detect and transmit all messages related to packet data transfer. The messages transmitted by a mobile station can be originated by the user's keyboard or by a separate data terminal connected to the station.

The invention is described above with a view to assembling a virtual channel without mentioning more closely on which particular radio channel the transmission of data packets will take place.

In accordance with the first embodiment, the radio channel reserved for packet data transmission is a traffic channel TCH normally used for transmitting speech. On termination of transmission said packet data channel is free for use of any other mobile station. Such first embodiment is described below.

Reference is made to FIG. 9 showing packet data transfer in mobile station originated mode. The figure is equivalent to FIG. 5 and the description thereof, with an additional remark that also a mentioning has been added therein on which channel each message is transmitted. So, a mobile station sends a packet data channel request to a base station using a common Random Access CHannel RACH, which all stations in the cell use when requesting a radio channel. The base station replies by a traffic channel assignment on the common Access Grant CHannel AGCH, whereafter the packet data transfer and acknowledgement of reception are carried out on the traffic channel. The paging transmitted on the Random Access CHannel RACH contains a value 001 in the "Establishment Cause" as in FIG. 3. Said channel paging request is a modification of a standard channel paging of the GSM system. The value "001" would mean that the direction of the packets is from the network to the mobile station. The purpose thereof is so that the value of the "Establishment Cause" field is different in the mobile originated case and the mobile terminated case is to ensure that the priority of the mobile terminated case is higher because the network has already been made to prepare a connection.

The network responds to the paging on the Access Grant CHannel AGCH with a message called "Packet Data Assignment". The message is a modification from the standard GSM message "Immediate Assignment". The modification is such that the bit configuration of the "message type" block of said standard message is 00111101 in the present invention, said configuration not being used for any other purposes in GSM. After the message "Packet Data Assignment" the signalling is not continued on the Stand-alone Dedicated Slow Control Channel SDCCH, as the case would be in standard traffic channel trafficking, but on a Faster Associated Control Channel. This should be included in the message sent to the mobile station. The standard message includes an information part "channel description" and it includes an element "channel type". This element informs that the traffic channel has to be connected with. The bit configuration illustrating the full rate traffic channel TCH and the control channel FACCH associated thereto is "00001". In this element also the timing advance TA and power control are transmitted, these being necessary data for the mobile station.

When the mobile station has received the above-described modified message, it immediately moves to the traffic channel and starts data packet transmission. If the assembly of the connection between the mobile phone and the Mobile Switching Center requires more signalling prior to transfer of the mobile phone to data packet transmission, the signalling can be carried out on the full rate control channel FACCH.

The operating time of the traffic channel can be limited relative to the time available or the number of packets. The simplest and most effective method is possibly to transmit all data from the transmission buffer and to release the traffic channel TCH after the buffer is empty. Since the reservation of a true channel takes a few hundreds of milliseconds, a timer can be provided in the telephone counting the time after emptying the buffer. The traffic channel is not released until a set time has elapsed, not immediately after the transmission of the last packet. So, a transmission can be repeated or more transmitted (if more data have been accumulated in the buffer) without setting up a channel. The use of a timer increases the sense of interaction because the channel need not be established again and again in each case. If the transmission rate of the packets is high, the timer keeps the traffic channel TCH continuously reserved and the user receives the replies immediately. The time setting of the timer can be set by the user.

The operator may also select one traffic channel only in the cell for transferring data packets or equally a great number or even all traffic channels.

FIG. 10 schematically shows the functions of the first embodiment when packet data are to be sent via the network from a mobile station. The only difference to the opposite case is that the network first informs the mobile station of the packet data transmission to come. The report takes place on the common paging channel PCH in a paging message. When the mobile station has received the paging, the activity is continued, as in the mobile station originated case, that is, the station transmits a channel request to the base station, and the operation goes on as described in association with FIG. 9.

FIG. 11 shows the formats of a packet data message of an arrangement in accordance with the first embodiment. The packets of the Virtual Channel Protocol, VCP, are produced using the OSI terminology in layer 3, above the link layer, and conducted via Layer 2 Relay Functions, L2R, to the Radio Link Protocol for transmission via the broadcast interface (radio path). The packet includes a header and a data part. The header includes the identification of the upper level protocol to be used. One of the upper level protocols is the protocol of the packet used in the signalling between the station and the Agent in association with the Mobile Switching Center. Other potential protocols are Internet Protocol (IP), Open Systems Interconnection (OSI) protocol and some fax protocols. The operator of the network may also add services of his own to be attended to by the Agent, these being provided with identifications of their own. The

header may alternatively be also provided with a field informing of the length of the packets. The length of the data part of the packets, or the number of higher level octets, varies. One packet can be transferred in one or more RLP frames.

A second embodiment of the invention is described, according to which packet data can be transmitted either on a traffic channel TCH or on a common Random Access CHannel RACH, using a channel request, which can be, as above, an 8 bit byte with a bit sequence "001" at the beginning. Thereafter, the network transmits on the Access Grant Channel, AGCH, a message requesting transmission of packet data, the message being a modification of the standard GSM message. The element determining the message type thereof includes bit configuration "00111101", indicating that a packet data case is in question. In block "Channel Type" the bit configuration "00001" indicates that the mobile station should move to the traffic channel TCH to transfer the packet data thereon, and the bit configuration "10000" indicates that it has to stay on the Random Access CHannel RACH and to transfer the data packets on that channel. The network makes a decision which channel is to be used. If the telephone traffic in the cell is large scale, the transmission is carried out on the traffic channel, but if it is minor, the Random Access CHannel RACH is used.

The duration of transmission on the Random Access CHannel RACH is limited by means of a timer or counter as the Timing Advance, TA, changes very rapidly and the channel reservation occupies the possibilities from the others to request for a connection to be formed.

FIG. 12 shows the formats of a packet data message of an arrangement in accordance with the second embodiment. Each frame is provided with an 8-bit ordinal number SN acting as identification of a connection. It is generated by the base station and transmitted to the mobile station in conjunction with the assignment message of the packet data. The identification is released after the connection ends. The identification is necessary so that the data included in the same connection with the random access channel and the traffic channel can be combined. FIG. 13 presents a case in which packet data are transmitted on a random access channel. On that channel the packet data are transmitted as standard bursts, and the figure shows the equivalence of the RACH channel frame as standard bursts.

The mobile station is enabled to present in the form of a wish, which of the channels it wants to use for transmitting data. Each TCH and RACH frame is provided with two command bits, informing the channel of the subsequent frame. The connection via the RACH channel can be discontinued if a request to move to the traffic channel TCH arrives, and likewise, the connection via the TCH channel can be discontinued if a request to move to the RACH channel arrives. The command bits C at the ends of the frames are available for use of the mobile station for a channel shift request and moreover, the termination of a data transfer can be reported therethrough. These two bits can therefore be used as follows:

bits "11"=move to the same channel

bits "01"=move to traffic channel TCH

bits "10"=switch to the common Random Access CHannel and the common Paging CHannel PCH

bits "00"=transmission over.

The switching onto the transmission channel can be implemented in two ways. After the switch-on-the-channel command transfer, the mobile station is allowed to request for a channel in a "packet data channel request" message and to wait for a channel assignment message to be able to select the channel on which the data traffic will take place. Another alternative is to read the message on the network side and if a channel switching is requested in the command bits, the "packet data assignment" message is sent without any "packet data request" message. The packet has been transferred to a plurality of RLP frames. One of the RLP frames is interlaced into 22 standard bursts of the TCH channel.

The transmission of data packets is described above in mobile station-originated mode in an instance according to the second embodiment. The instance in which packet data are transmitted via the network to a mobile station differs from the above-mentioned case only in that the network reports the mobile station of future transmission in a particular "packet data paging request" message which it sends on a common Paging CHannel. The message is a modification of the paging of the GSM system being provided with a free bit configuration for this purpose. "001000011" is selected for the bit configuration. As an extension. a data field is added in the message, wherein the data to be transmitted to the user is transferred. After receiving this paging message (or a series of paging messages including the packet), the mobile station opens a connection and acknowledges the packet.

FIG. 14 presents schematically the events in temporal order when transmitting data packets via the network to the mobile station. The packet transfer first takes place on a traffic channel, moves onto a random access channel, returns on the traffic channel and then on the random access channels. On the random access channels the trafficking time runs out, and the connection is terminated forcedly.

In view of the foregoing description it will be evident to a person skilled in the art that various modifications may be made within the scope of the invention.

The scope of the present disclosure includes any novel feature or combination of features disclosed therein either explicitly or implicitly or any generalisation thereof irrespective of whether or not it relates to the claimed invention or mitigates any or all of the problems addressed by the present invention. The applicant hereby gives notice that new claims may be formulated to such features during prosecution of this application or of any such further application derived therefrom.

What we claim is:

1. A radio telephone system comprising:

a mobile station having means for storing a first parameter, relating to another station, for setting up a real data communication channel having a reserved physical path between said mobile station and said another station; and

said another station comprising a fixed station having means, wherein a second parameter relating to the mobile station for setting up said real data communication channel is capable of being stored, for forming a virtual data communication channel between the mobile station and the fixed station, said virtual data communication channel being a non-physical registration relationship using said first and second parameters and lacking a reserved path between said mobile station and said fixed station preparatory to the establishment of said real data communication channel, whereby the establishment of a real data communication channel is expedited when data is to be communicated between said mobile and fixed stations.

2. A system according to claim 1, further comprising means for communicating a first parameter relating to said mobile station to be stored in the fixed station parameter storing means to form the virtual data communication channel when a mobile station having data communication capability registers with the fixed station.

3. A system according to claim 1, further comprising means for forming the real data communication channel when a mobile station registered with the fixed station requests the setting up of a data communication channel.

4. A system according to claim 1, further comprising means for forming traffic channels, usually reserved for speech communication, and control channels between the mobile station and the fixed station, and wherein the real data communication channel is a traffic channel usually reserved for speech communication.

5. A system according to claim 1, further comprising means for forming traffic channels, usually reserved for speech communication, and control channels between the mobile station and the fixed station, and wherein the real data communication channel is a control channel.

6. A system according to claim 1, further comprising control means for controlling communication between the mobile station, the fixed station and an external communication network.

7. A system according to claim 1, further comprising means for adapting the real data communication channel for transmitting packet data.

8. A digital time-division cellular network having a base station and a plurality of mobile stations, wherein radio channels between the mobile stations and the base station comprise:

a plurality of Traffic CHannels (TCH) for transferring speech and data between the mobile stations and the base station,

a control channel in association with each of said TCH channels, said control channels comprising:

a Random Access CHannel (RACH), for conducting signals from the mobile stations to the base station requesting a TCH channel,

a common Paging CHannel (PCH), for conducting a paging signal from the base station to a mobile station,

an Access Grant CHannel (AGCH), for conducting a signal from the base station to inform a mobile station of the TCH channel assigned thereto,

and further comprising:

a packet data service unit (Agent) for connecting the cellular network to a data network,

means for switching a mobile station to the Agent and signalling for the setting up of a connection to the data network,

means in a mobile station and the Agent for storing a number of parameters relating to each other, said parameters including a Radio Link Protocol (RLP) and forming a virtual channel, said virtual channel being a non-physical registration relationship using said parameters and lacking a reserved path between said mobile station and said Agent preparatory to the establishment of a packet data transfer channel,

means, responsive to a request from a mobile station to transfer or receive data packets, for assembling a packet data transfer channel between the mobile station and the Agent, making use of the parameters of the virtual channel, and wherein said packet data transfer channel comprises a first part comprising a radio channel between said mobile station and the base station and a second part comprising a time slot in a digital trunk line between said base station and said Agent,

means for transferring data packets over said packet data transfer channel, means, responsive to the termination of data packet transfer over said packet data transfer channel, for disassembling at least said radio channel, and

means for maintaining the virtual channel until the release of a mobile station from the Agent.

9. A network according to claim 8, wherein the Agent comprises:

means for registering all mobile stations connected to the Agent,

means for informing a mobile station of any data packets addressed thereto,

means for transferring data packets from a mobile station addressed to the data network,

means for transferring messages to the mobile station entering the Agent from the data network,

means for buffering data packets,

means for performing encrypting/decrypting,

means for performing compression/decompression of the data to and from a mobile station,

means for updating a data base of the location of the mobile stations,

means for receiving data packets from the data network addressed to the cellular network and transferring them to the mobile stations, and

means for removing a mobile station from the register after it is disconnected from the Agent.

10. A network according to claim 8, wherein the packet data service unit (Agent) comprises means for adapting the data packets from the data network to virtual channel protocol packets, said virtual channel protocol packets being composed of one or more radio link traffic protocol (RLP) frames.

11. A network according to claim 10, wherein the virtual channel protocol packets comprise an identification part indicating whether the contents of a packet contain signalling data or upper layer data.

12. A network according to claim 8, further comprising means, in a mobile station, for initiating the transmitting of data packets by sending a request on the RACH channel for establishing a packet data transfer channel, said request being a modification of the standard channel establishing request of the cellular network.

13. A network according to claim 8, further comprising means in the base station, responsive to data packets to be transferred to a mobile station, for sending a message on the common PCH channel to the mobile station about said data packets to be transferred and means at the mobile station, responsive to said message for sending on the RACH channel a request to the base station for establishing a packet data transfer channel, said request being a modification of the standard channel establishing request of the cellular network.

14. A network according to claim 8, further comprising means, in the base station, for transmitting control channel data used in channel establishment signalling and packet data transfer channel data to the mobile stations.

15. A network according to claim 14, further comprising means for establishing, after the channel establishment signalling between the mobile station and the base station, said second part of the packet data transfer channel with said Agent, whereby the entire packet data transfer channel is ready for packet transfer.

16. A network according to claim 8, wherein said cellular network comprises a Dedicated Fast Access Channel (FACCH), and further comprising means for causing said

channel establishment signalling to be carried out on a Dedicated Fast Access CHannel, FACCH of the cellular network.

17. A network according to claim 8, further comprising a base station controller connected to the base station and wherein the second part of the packet data transfer channel is a direct PCM connection from the base station controller to the Agent, whereby said second part of the packet data transfer channel is active irrespective of the packet data transfer.

18. A network according to claim 8, wherein the second part of the packet data transfer channel is a variable time slot on the PCM trunk line, and further comprising means whereby said second part is disassembled after the termination of the data packet transfer.

19. A network according to claim 8, wherein the first part of the packet data transfer channel is a TCH channel.

20. A network according to claim 8, further comprising means for causing the first part of the packet data transfer channel to be a RACH channel when packet data are transferred from the mobile station to the Agent, and to be the common PCH channel when packet data are transferred from the Agent to the mobile station.

21. A network according to claim 8, further comprising means for causing, in the course of a transfer of packet data, the first part of the packet data transfer channel to be any one of the TCH channel, the RACH channel, and the common PCH channel.

22. A network according to claim 8, wherein a broadcast paging message transmitted to all mobile stations of the cellular network and a multicast paging message transmitted to certain mobile stations of the cellular network are transmitted on the data section of the broadcast and the multicast protocols, respectively.

23. A method of transmitting data in a radio telephone network comprising:

storing at a fixed station a first parameter relating to a mobile station and the setting up of a real data communication channel between the mobile station and the fixed station; and

storing at the mobile station a second parameter relating to the fixed station and the setting up of a real data communication channel between the mobile station and the fixed station, for forming a virtual data communication channel between the mobile station and the fixed station, the virtual data communication channel being a non-physical registration relationship using the first and second parameters and lacking a reserved path between the mobile station and the fixed station preparatory to the establishment of a real data communication channel, thereby expediting establishment of a real data communication channel between the mobile station and the fixed station in response to an appropriate request.

24. A method according to claim 23, further comprising forming the virtual data communication channel when a mobile station having data communication capability registers with the fixed station.

25. A method according to claim 23, further comprising forming the virtual data communication channel when a mobile station registered with the fixed station requests the setting up of a real data communication channel.

26. A method according to claim 23, wherein said network comprises traffic channels usually reserved for speech communication, and control channels, and further comprising utilising a traffic channel usually reserved for speech communication for the real data communication.

27. A method according to claim 23, wherein said network comprises traffic channels usually reserved for speech communication, and control channels, and further comprising utilising a control channel as the real data communication channel.

28. A radio telephone having storage means therein to store a parameter relating to a fixed station for setting up a real data communication channel with said fixed station and for forming a virtual data communication channel with the fixed station by utilizing a parameter relating to said radio telephone stored in said fixed station, the virtual data communication channel being a non-physical registration relationship using the respective stored parameters and lacking a reserved path between the radio telephone and the fixed station preparatory to the establishment of a real data communication channel, thereby expediting establishment of a real data communication channel between said radio telephone and said fixed station in response to an appropriate request.

29. A radio telephone according to claim 28, further comprising means adapted to store said parameter in said storage means when said radio telephone has a data communication ability and when said radio telephone registers with the fixed station.

30. A radio telephone according to claim 28, further comprising means adapted to store said parameter in said storage means when said radio telephone is registered with the fixed station and requests the setting up of a data communication channel.

31. A radio telephone system according to claim 1 wherein said mobile station comprises a radio telephone.

32. A method according to claim 23 wherein said mobile station comprises a radio telephone.

* * * * *

EXHIBIT B

US006359904B1

(12) **United States Patent**
Hämäläinen et al.

(10) **Patent No.: US 6,359,904 B1**
(45) **Date of Patent: Mar. 19, 2002**

(54) **DATA TRANSFER IN A MOBILE TELEPHONE NETWORK**

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(73) Assignee: **Nokia Mobile Phone Ltd.**, Espoo (FI)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

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(30) **Foreign Application Priority Data**

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(51) **Int. Cl.**[7] **H04J 3/16**; H04J 3/22

(52) **U.S. Cl.** **370/469**; 370/328; 455/422

(58) **Field of Search** 370/328, 329, 370/336, 337, 345, 347, 465, 469; 455/422, 450

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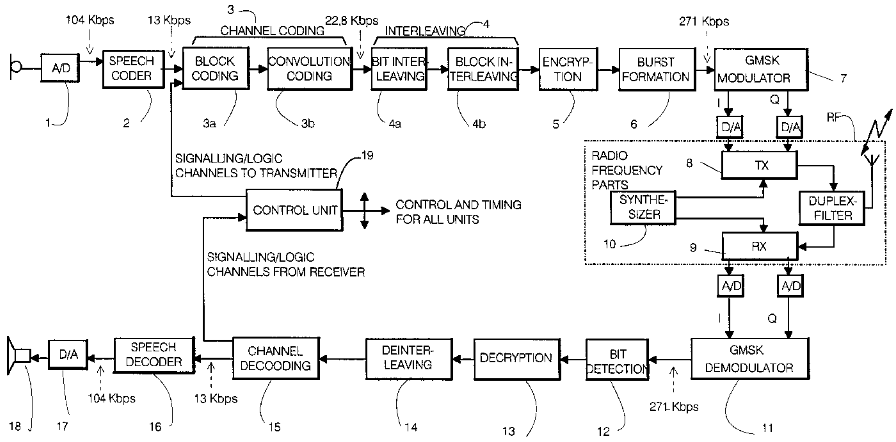
Primary Examiner—Ajit Patel
Assistant Examiner—Bob A. Phunkulh
(74) *Attorney, Agent, or Firm*—Perman & Green, LLP

(57) **ABSTRACT**

The scope of the present invention is a method for data transfer in a digital mobile communications system, in which method it is handled data in certain layers according to certain protocols, in a certain layer out of said layers it is transferred user data in radio blocks (RB) over a physical radio channel between a mobile station and a fixed mobile communications network, for the transfer of said certain layer it is formed in the radio block (RB) a payload of a certain size comprising check bits (CHB) connected with the performing of the transfer and transfer bits (TB) available for the transfer of user data, each radio block (RB) is channel coded using a certain coding method and the size of said payload is dependent on the coding method. In the transfer bits (TB) of the radio block to be coded using at least a certain coding method it is transferred user data in a first part of the transfer bits and in a second part of the transfer bits it is transferred fill bits in such a way that it is chosen for the transfer of user data a number of transfer bits divisible by eight.

12 Claims, 3 Drawing Sheets



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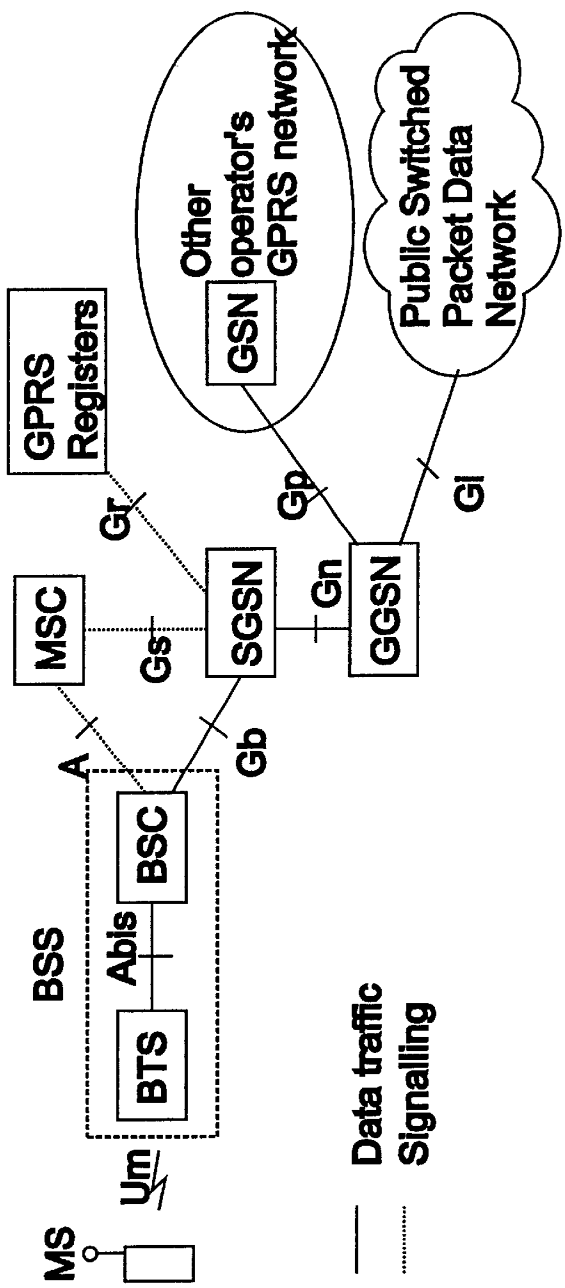
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Um
BSS
BTS
Abis
BSC
A
MSC
Gb
Gs
SGSN
Gr
GPRS Registers
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GGSN
Gp
GSN
Other operator's GPRS network
Gi
Public Switched Packet Data Network
Data traffic
Signalling

Fig. 1

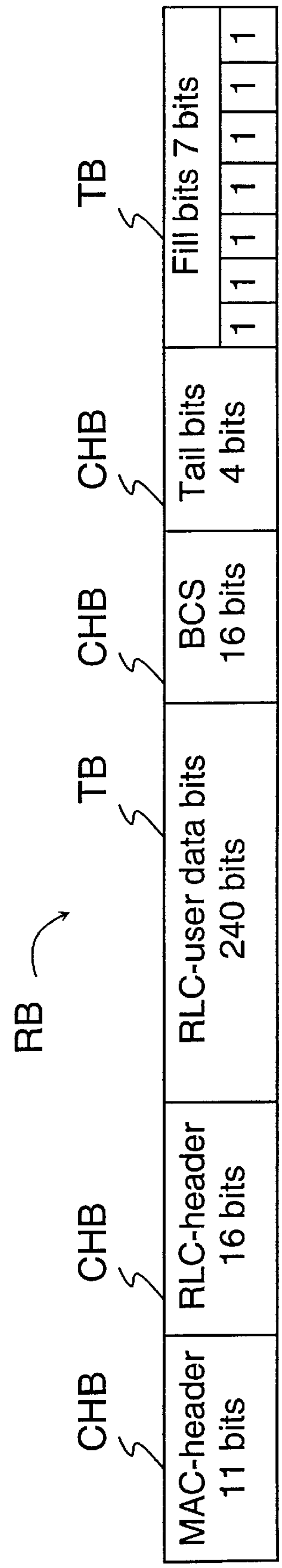

RB
CHB
MAC-header 11 bits
CHB
RLC-header 16 bits
TB
RLC-user data bits 240 bits
CHB
BCS 16 bits
CHB
Tail bits 4 bits
TB
Fill bits 7 bits
1 1 1 1 1 1 1

Fig. 4

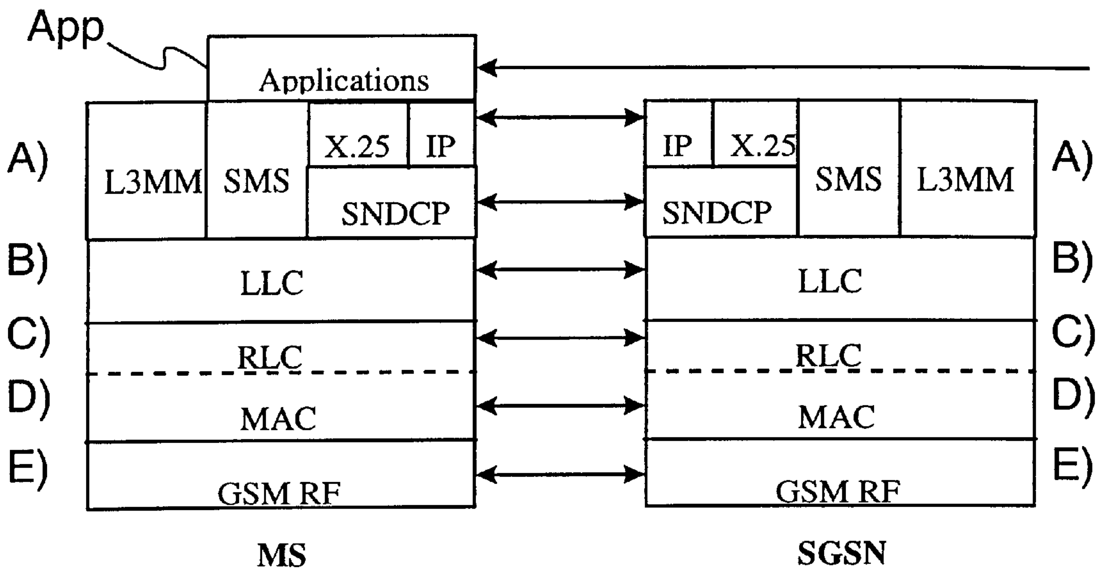


Fig. 2a

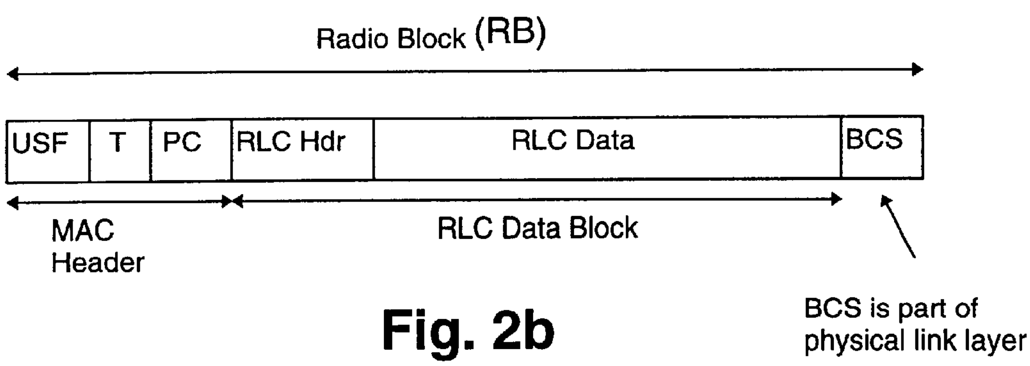


Fig. 2b

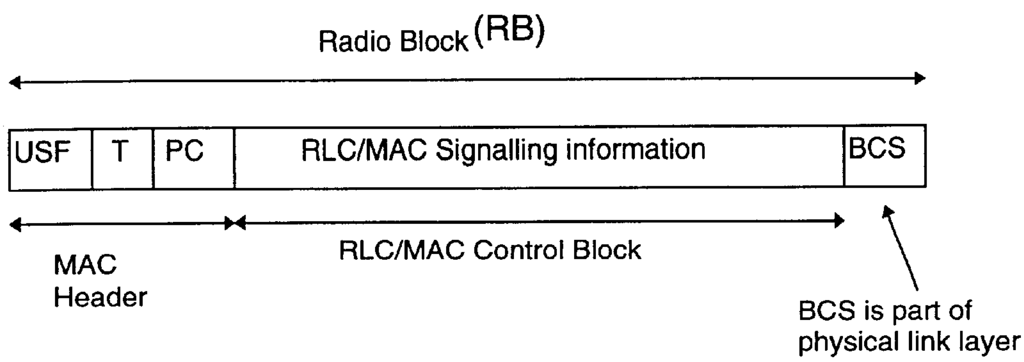
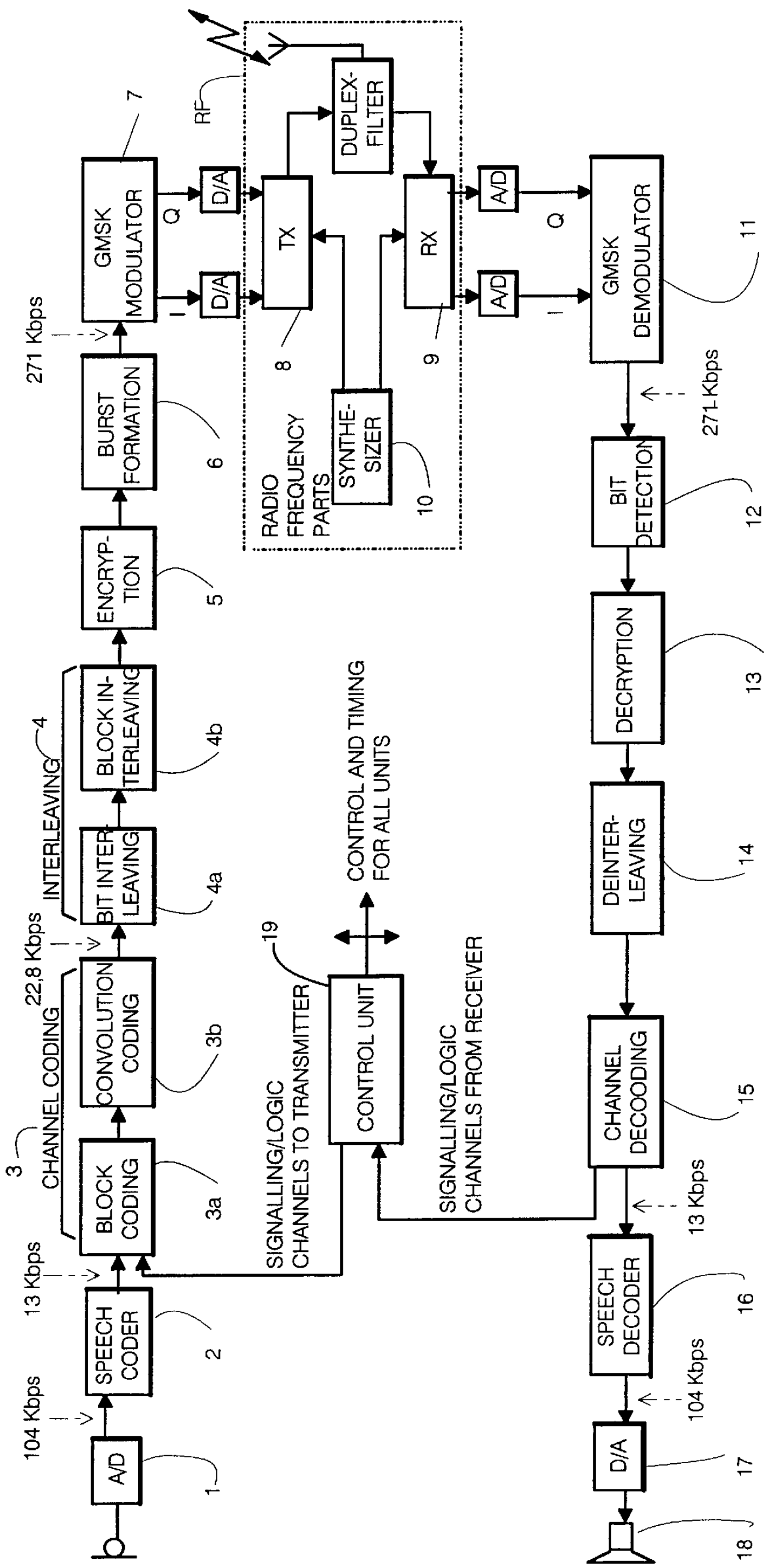


Fig. 2c


A/D
104 Kbps
SPEECH CODER
13 Kbps
BLOCK CODING
CONVOLUTION CODING
CHANNEL CODING
22,8 Kbps
BIT INTER-LEAVING
BLOCK IN-TERLEAVING
INTERLEAVING
ENCRYP-TION
BURST FORMATION
271 Kbps
GMSK MODULATOR
I
Q
D/A
D/A
TX
RX
DUPLEX-FILTER
RF
SYNTHE-SIZER
RADIO FREQUENCY PARTS
A/D
A/D
GMSK DEMODULATOR
271 Kbps
BIT DETECTION
DECRYPTION
DEINTER-LEAVING
CHANNEL DECOODING
13 Kbps
SPEECH DECODER
104 Kbps
D/A
CONTROL UNIT
CONTROL AND TIMING FOR ALL UNITS
SIGNALLING/LOGIC CHANNELS TO TRANSMITTER
SIGNALLING/LOGIC CHANNELS FROM RECEIVER
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3b
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4a
4b
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FIG. 3

DATA TRANSFER IN A MOBILE TELEPHONE NETWORK

FIELD OF THE INVENTION

The present invention relates to a method for data transfer in a digital mobile communications network, in which method it is handled user data in certain layers according to certain protocols, in a certain layer of said layers it is transferred user data over a physical radio channel between a mobile station and a fixed mobile communications network in radio blocks, for the transfer of said layer it is formed in the radio block a payload of a certain size comprising check bits connected with the performing of the transfer and transfer bits available for the transfer of user data, each radio block is channel coded using a certain coding method and the size of said payload is dependent on the coding method. The invention also relates to a transmitter/receiver device operating according to the method and a mobile communications system. The invention relates in particular to data transfer in the GSM-system in the GPRS-packet switched service.

BACKGROUND OF THE INVENTION

Out of the present mobile communications systems a majority is offering data- and voice services based upon circuit switched technique. In the circuit switched technique a transfer connection is maintained during the whole connection even if no information would be transferred time to time. This unnecessarily consumes the transfer resources, shared by also many other users, in which case the maintaining of a circuit switched transfer connection to one user consumes unnecessarily the transfer resources of other users. Because of the bursts in the GSM-transmission, data services are not optimal in the circuit switched technique. However, the packet switched information transfer for the increasing of the efficiency of the utilization of a channel is known.

As well as the fixed network also a future mobile communications network must be able to transfer both circuit switched and packet data transfer, e.g. ISDN-transfer (Integrated Services Digital Network) and ATM-transfer (Asynchronous Transfer Mode). For information transfer using packet switching it is known in mobile communications systems a protocol based upon packet reservation multiple access called PRMA (PRMA, Packet Reservation Multiple Access). It is also spoken of as packet radio. PRMA is a technique for multiplexing digital speech or data in a time division carrier wave, i.e. PRMA uses in a radio channel a time division multiple access (TDMA, Time Division Multiple Access), in which case transmission and reception take place at certain moments using time division. The PRMA-protocol has been developed to utilize the discontinuity of speech transfer in order to support more users than the number of speech channels in a time division carrier wave. In such a case a channel is allocated to a mobile station, for example a speech channel when speech is being produced and when the speaking ends the channel is released, in which case the mobile station does not unnecessarily reserve capacity, but the channel is channel is free for other purposes, for example the transmissions of other mobile stations in the cell. The PRMA-protocol is used in cellular mobile communications systems in the communication between a mobile station and a base station. The GSM GPRS (General Packet Radio Service)-system is an example of a system based upon a PRMA-type protocol.

GPRS is a new GSM-service, by using which the packet radio operation can be made available to GSM-users. GPRS

reserves radio resources only when there is something to transmit, in which case the same resources are shared between all mobile stations as needed. The normal circuit switched network of the GSM-system has been designed for circuit switched speech transmissions. The main goal of the GPRS-service is to realize the connection from a mobile station to a public data network using prior known protocols, such as TCP/IP and X.25. However, there is a connection between the packet switched GPRS-service and the circuit switched services of the GSM-system. In a physical channel resources can be reused and certain signalling can be common to both. It is possible to reserve in the same carrier wave time slots for circuit switched use and for the packet switched GPRS-use.

FIG. **1** presents telecommunication network connections in a packet switched GPRS-service. The main element of the infrastructure of the network for GPRS-services is a GPRS-support node, so called GSN (GPRS Support Node). It is a mobility router which realizes the connecting and co-operation between different data networks, for instance to PSPDN (Packet Switched Packet Data Network) through interface Gi or to another operator's GPRS-network through interface Gp, mobility management using GPRS-registers over interface Gr and the transfer of data packets to mobile stations MS independent of their location. It is possible to integrate physically GPRS-node GSN with mobile switching center MSC (Mobile Switching Center) or it can be a separate network element based upon the architecture of data network routers. User data passes directly between support node GSN and base stations system BSS, consisting of base stations BTS and base station controllers BSC, through interface Gb, but between support node GSN and mobile switching center MSC there is signalling interface Gs. In FIG. **1** the uninterrupted lines between blocks represent data traffic (i.e. the transfer of speech or data in a digital form) and the interrupted lines represent signalling. Physically the data can pass transparently over mobile switching center MSC. The radio interface between mobile station MS and the fixed network passes through base station BTS and it has been marked with reference Um. References Abis and A represent the interface with base station BTS and base station controller BSC, and respectively between base station controller BSC and mobile switching center MSC, which is a signalling connection. Reference Gn represents an interface between the different support nodes of the same operator. The support nodes are normally divided into gateway support nodes GGSN (Gateway GSN) and serving or home support nodes SGSN (Serving GSN) as presented FIG. **1**.

The GSM-system is a time division multiple access-type (TDMA, Time Division Multiple Access) system, in which the traffic in the radio path is time-divided and takes place in repeated TDMA-frames, each of which consists of several (eight) time slots. In each time slot it is transmitted an information packet in form of a radio frequency burst of finite duration consisting of a number of modulated bits. The time slots are mainly used as control channels and traffic channels. On the traffic channels it is transferred speech and data and in the control channels it is carried out signalling between base station BTS and mobile station MS.

In the following it is explained the protocols of GPRS and the protocol hierarchy in radio interface Um between mobile station MS and a fixed network (home support node SGSN) with reference to FIG. **2***a*. User data is handled hierarchically on different levels, when it is converted into a form suited for the physical radio path and the public data network. On the highest level A) the user data (coming e.g.

from an application App) is in a form suited for the protocol of the public data network, such as TCP/IP and X.25 and on the lowest level E) the data is in a form suited for transferring in the GSM-radio path.

The highest level A) protocol SNDCP (Subnetwork Dependent Convergent Protocol), i.e. a convergence protocol dependent of a subnetwork is explained in more detail in GSM radio specifications 04.65 and 03.60. According to SNDCP a network protocol data unit is segmented between mobile station MS and home support node SGSN into one or several SNDCP data units, the maximum size of the payload of which is approximately 1600 octets. The SNDCP-data unit is transferred in one LLC-fame (Logical Link Control) over the radio interface. The SNDCP-protocol includes multiplexing of user data, segmenting and compressing, and the compressing of the TCP/IP-header. It is possible to transfer in the SNDCP-protocol different network level protocols, such as IP, X.25, PTM-M and PTM-G. The size of a SNDCP user data field is, as to the total number of bits, divisible by eight bits, i.e. it is octet oriented.

The protocol of the next B) level, the LLC-protocol or the logical link control protocol has been explained in more detail in GSM standard specifications 04.64 and 03.60. The LLC-protocol provides a reliable logical link between a mobile station and home support node SGSN. SNDCP-, short messages and GPRS signalling messages are transmitted in LLC-frames which have a frame header containing numbering and a temporary address field, an information field of variable length and a frame check sequence. The functionality of LLC includes maintaining the communication context of mobile station MS and home support node SGSN, the transmitting of acknowledged and unacknowledged frames, the detection and retransmitting of corrupted frames. LLC-frames are transmitted in one or several radio blocks. The logical link is maintained when mobile station MS moves between cells within the area of one home support node SGSN. If mobile station MS moves into the area of another home support node SGSN, a new logical link must be established. The size of a LLC-protocol user data field is, as to the total number of bits, divisible by eight bits, i.e. it is octet oriented.

The next level C) after LLC, the RLC-level (Radio Link Control) has been explained in more detail in GSM standard specifications 03.64. The LLC-frame is being transmitted continuously. The variable length LLC-frames is transmitted in one or more RLC-blocks. The functionality of RLC between mobile station MS and home support node SGSN is to detect the corrupted RLC-blocks and to ask for a selective retransmission of the corrupted blocks. A retransmission request comprises a bit map indicating each air path block to be either corrupted or successfully received. Based upon the bit map the transmitter retransmits the corrupted blocks. The total size of an RLC-block is, the header and user data included, as to the number of bits, is divisible by eight bits, i.e. it is octet oriented.

Also level D), the MAC-level (Medium Access Control) has been explained in more detail in GSM standard specifications 03.64. MAC is used for dividing radio channels between mobile stations and for the allocating of a radio channel for a mobile station for transmission and reception as needed. The functionality of MAC includes a separate header containing uplink state flag USF (Uplink State Flag), block type indicator T and eventual power control information PC (Power Control). The MAC-header and the RLC-data block are placed in radio block RB (see FIGS. **2***b* and **2***c*) to be transmitted on the physical layer.

Protocol level E) describes the physical layer or GSM-radio path, in which messages are transferred in radio blocks RB presented in FIGS. **2***b* and **2***c*. Radio block RB includes a MAC-header, an information part containing the data or the signalling (RLC-data block, FIG. **2***b* or an RLC/MAC-signalling-information block, FIG. **2***c*) and block check sequence BCS (Block Check Sequence). Each radio block is interleaved in four standard bursts. Before the interleaving it is performed a channel coding on the radio block. For the channel coding there are four different coding schemes CS-**1**, CS-**2**, CS-**3** and CS-**4** (Coding Scheme). A mobile station must support all four alternatives. In the channel coding a convolutional coding is performed on the information part. A pre-coding is performed on uplink state flag USF (Uplink State Flag), in which case the length of USF after the pre-coding is dependent on the channel coding method CS-**1** . . . CS-**4** used. After the channel coding the size of the radio block is according to the GSM-specification 456 bits. Prior to the convolutional coding the payload according to the different coding method varies, and an octet oriented data stream is not achieved with all coding methods CS-**1** . . . CS-**4**. Only CS-**1** produces an octet oriented data stream, but the other the channel coding methods CS-**2** . . . CS-**4** do not do it according to the present protocols. This hampers data stream between different layers A)–E) in mobile station MS and in the mobile communications network, i.e. in base station system BSS and in home support node SGSN.

SUMMARY OF THE INVENTION

Now it is introduced such a method, with which data flow can be made simpler between all hierarchy levels or between mobile station MS and the different protocols of mobile communications network BSS; SGSN. This is achieved by bringing the user data flow into octet form on all protocol levels of the GPRS-service, in particular on the lower levels, by setting a certain number of bits as fill bits instead of using them for the transfer of user data. With this method it is possible to make the payload of a radio block octet oriented when any of channel coding methods CS-**1** . . . CS-**4** is used. A certain number of the bits of radio block RB, the being determined according to the method, are set prior to channel coding and the interleaving of the radio block (in four bursts) to transfer fill bits in such a way that the number of bits in the radio block transferring user data is divisible by eight prior to the channel coding. By using the method the handling of data, in particular that of the user data to be transferred, is made octet oriented on all GPRS-protocol levels. Because the radio block is made octet oriented the operation can after the channel coding be carried out fully in accordance with the GSM-specifications.

If this method were not used, the transmissions of two radio blocks would be mixed in such a way that the last bits of the preceding radio block would be transferred in the same burst with the first bits of the next radio block. This would make the handling of the data and protocols, and the equipment executing them difficult when octets coming from a higher protocol level should be distributed to different blocks on lower protocol levels.

The method according to the invention is characterized in that in the transfer bits of a radio block coded using at least a certain coding method it is transferred user data in a first part of the transfer bits and fill bits in a second part in such a way that it is chosen such a number bits for the of transfer user data which is divisible by eight.

The transmitter/receiver device according to the invention is correspondingly characterized in that it comprises control means for transferring user data in a first part of radio block transfer bits coded using at least a certain coding method and for transferring fill bits in a second part of said transfer bits, and said first part of transfer bits comprises a number of bits divisible by eight.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention is described in detail in the following with reference to enclosed figures, of which

FIG. **1** presents the structure of a telecommunication network in the GSM GPRS-packet service data transfer,

FIG. **2***a* presents the different protocols levels of the GPRS-service,

FIG. **2***b* presents a radio block to be transferred in the radio interface,

FIG. **2***c* presents another radio block to be transferred in the radio interface,

FIG. **3** presents the block diagram of a GSM-system transceiver,

FIG. **4** presents a radio block according to the invention to be transferred in the radio interface.

DETAILED DESCRIPTION

In order to illustrate the handling of a transmitter/receiver and a physical layer according to the invention, it is explained in the following the transmitter- and receiver function of the GSM-system with reference to FIG. **3**, in which it is presented a block diagram of a transmitter/receiver in a mobile telephone according to the GSM-system. The transmitter/receiver of a base station differs from the transmitter/receiver of a mobile telephone usually in the respect that it is a multi-channel one and it has no microphone nor loudspeaker, in other respects it is in structure and operating principle similar to the transmitter/receiver of a mobile telephone.

The first stage of a transmission sequence is digitizing **1** of analogue speech and encoding **2**. Sampling with A/D-converter **1** is carried out a 8 kHz frequency and the speech encoding algorithm assumes the input signal to be 13 bit linear PCM. The samples obtained from the A/D-converter are segmented into 160-sample speech frames, in which case the duration of each speech frame is 20 ms. Speech encoder **2** handles 20 ms speech frames, i.e. prior to the commencing of the encoding it is taken 20 ms of speech in a buffer. The coding operations are performed frame by frame or on their subframes (in 40-sample blocks). As a result of the encoding by speech encoder **2** it is obtained 260 bits out of one frame.

After speech encoding **2** it is performed channel coding **3** for example in two stages depending on the coding method used, when at first one part of the bits (e.g. 50 most significant of 260 bits) are protected using block code **3***a* (=CRC, 3 bits) and after that these and the next most important bits (132) are further protected using convolutional code **3***b* (coding ratio 1/2) ((50+3+132+4)*2=378, and a part of the bits are taken unprotected (78). As presented in FIG. **3**, signalling- and logical messages and the data to be transmitted come directly from control unit **19** controlling the blocks of the telephone to block coding block **3***a*, and thus naturally no speech encoding is performed on these data messages. Correspondingly, signalling- and logical messages and the received data are taken from channel decoding block **15** to control unit **19**. In block encoding **3***a* a bit string is attached at the end of a speech frame, using which bit string it is possible to detect transfer errors at reception. In convolutional coding **3***b* it is increased the redundancy of a speech frame. All in all, a total of 456 bits per each 20 ms frame is transmitted.

These 456 bits are interleaved **4** and also interleaving **4** is performed in two stages. At first **4***a* the order of bits is mixed and the mixed bits are divided into eight blocks of equal size. These blocks are further distributed **4***b* into eight subsequent TDMA-frames, in which case the interleaved 456 bits are transmitted in eight time slots of the radio path (57 bits in each). With the interleaving it is striven for to spread transfer errors, which usually occur as error bursts, evenly over all the data to be transmitted, in which case the channel decoding operates at its most effective. After the deciphering of the interleaving an error burst is converted into individual error bits which can be corrected in the channel decoding. The following stage in the transmission sequence is the ciphering **5** of data. Ciphering **5** is carried out using an algorithm which is one of the most guarded secrets of GSM. With the ciphering it is striven for to prevent any unauthorized listening of calls.

Out of the ciphered data it is formed **6** a bust to be transmitted by adding in it a learning sequence, tail bits and a protection time. The burst to be transmitted is brought to GMSK-modulator **7** which modulates the burst for transmission. The GMSK-modulation method (Gaussian Minimum Shift Keying) is a digital, constant amplitude modulation method, in which the information is contained in the shifts of phase. Transmitter **8** mixes the modulated burst through one or more intermediate frequencies into 900 MHz and transmits it through an antenna to the radio path. Transmitter **8** is one of three radio frequency blocks RF. Receiver **9** is the first block on the reception side and it performs the operations inverted to those of transmitter **8**. The third RF-block is synthesizer **10** which takes care of the forming of frequencies. In the GSM-system it is used frequency jumping, in which transmission- and reception frequencies are changed in each TDMA-frame. The frequency jumping improves the quality of the connection, but sets strict requirements on synthesizer **10**. Synthesizer **10** must be capable of moving from one frequency to another very quickly, in less than one millisecond.

In reception it is carried out operations inverted to transmission. After RF-receiver **9** and demodulator **11** it is carried out bit detection **12** using for example a channel correction unit, in which bits are detected from the received samples i.e. it is tried to find out the transmitted bit sequence. After the detection ciphering **13** and interleaving **14** are deciphered and channel decoding **15** is performed on the detected bits and the check sum is checked using a cyclic redundance check (CRC, Cyclic Redundance Check). In channel decoding **15** it is striven for to correct the bit errors occurred at the transfer of the burst. In a 260 bit speech frame after channel decoding **15** there are the transmitted parameters representing the speech, by using which speech decoder **16** forms the digital samples of the speech signal. The samples are D/A-converted **17** for reproduction with loudspeaker **18**.

In a transmitter/receiver as the central control unit of a mobile station there is control unit **19** which essentially controls all blocks **1–18** and coordinates their operations and controls timing. Control unit **19**usually comprises for example a microprocessor. The protocols according to hierarchy level A)–D), presented in FIG. **2***a*, are executed preferably in control unit **19** and the processing of user data to the physical channel (in transmission beginning from channel coding an in reception until channel decoding) is performed in blocks **3–15**.

For channel coding **3** there are four different coding schemes CS-**1**, CS-**2**, CS-**3** and CS-**4** (Coding Scheme). A mobile station must support each method. The data rates of these coding methods are 9.05, 13.4, 15.6 and 21.4 kbps respectively. Coding method CS-**1** contains a convolutional coding having a coding ratio of 1/2, which is used in the GSM-system on the SDCCH-channel. In coding methods CS-**2** and CS-**3** it is also at first performed a convolutional coding having a coding ratio of 1/2, after which fill bits are removed by puncturing in order to achieve the desired 456 bits. Coding method CS-**4** has no FEC-error protection (Forward Error Protection), i.e. no convolutional coding is performed on the data.

In the following it is explained in more detail the channel coding carried out in a packet data traffic channel (PDTCH, Packet Data Traffic Channel). Radio block RB presented in FIG. **2***b*, in which the RLC-data block is transferred, can be coded using one of the above channel coding methods CS-**1** . . . CS-**4**, while radio block RB presented in FIG. **2***c*, in which the RLC/MAC-control block is transferred is always coded using channel coding method CS-1.

In the first stage of coding it is added at the end of a radio block a block check sequence BCS (Block Check Sequence) for error detection. After this in coding methods CS-**1** . . . CS-**3** it is performed on the uplink status flag or USF a pre-coding (except in method CS-**1**), four tail bits are added and convolutional coding is performed according to above description, and puncturing in methods CS-**2** and CS-**3** in order to achieve the desired coding rate (456 bits).

The coding parameters of the different methods are presented below in Table 1.

TABLE 1

Scheme	Code rate	USF	Precoded USF (a)	Radio Block excl. USF and BCS (b)	BCS (c)	Tail (d)	Coded bits (e)	Punctured bits (f)	Data rate kb/s
CS-1	½	3	3	181	40	4	456	0	9.05
CS-2	≈⅔	3	6	268	16	4	588	132	13.4
CS-3	≈¾	3	6	312	16	4	676	220	15.6
CS-4	1	3	12	428	16	—	456	—	21.4

Table 1 shows that the length of USF after pre-coding/bit processing is 3, 6, 6 and 12 bits respectively in the different methods (column a). Block check sequence BCS is 40 bits in CS-**1** method and in the other methods 16 bits (column c). After convolutional coding **3***b* with code rate 1/2 it is obtained 456, 588 and 676 coded bits in methods CS-**1** . . . CS-**3** and in method CS-**4** it is obtained directly 456 bits without convolutional coding (column e). By adding together the bits in columns a–d it is obtained the payload according to each method. It is then seen that the payload in methods CS-**1**, CS-**2** and CS-**3** is 228, 294 and 338 bits and the number of bits is doubled in the convolutional coding in accordance with column e. In method CS-**4** it is obtained a payload of 456 bits. When it is known that the length of a pre-coded USF varies 3–12 bits and that the total length of T and PC is 5 bits, it is obtained as the size of a MAC-header field 8, 11, 11 and 17 bits. The number of tail bits is 4 in methods CS-**1** . . . CS-**3** and 0 in method CS-**4**. In this way the number of bits available for the transfer of the other data is as presented in Table 2.

TABLE 2

Payload bits	MAC header	BCS	Tail bits	Remaining bits
CS-1: 228 −	8 −	40 −	4 =	176
CS-2: 294 −	11 −	16 −	4 =	263
CS-3: 338 −	11 −	16 −	4 =	307
CS-4: 456 −	17 −	16 −	0 =	423

In these bits it is transferred the RLC-header and the RLC-data containing the actual user data. These are in the RLC-layer (layer C in FIG. **2***a*) divisible by eight. In order to keep the handling and transfer of user data octet oriented according to the invention, two octets or 16 bits are reserved for the header field, and the number of RLC-data block bits shown in Table 3 is transferred, when in certain methods bits are left over for user data transfer.

TABLE 3

Payload bits	MAC header		RLC header		RLC data bits		BCS		Tail bits		Additional bits
CS-1: 228 =	8	+	16	+	160	+	40	+	4	+	0
CS-2: 294 =	11	+	16	+	240	+	16	+	4	+	7
CS-3: 338 =	11	+	16	+	288	+	16	+	4	+	3
CS-4: 456	17	+	16	+	400	+	16	+	0	+	7

According to the invention it is not user data that is transferred in these additional bits of a radio block but fill bits in order to arrange the handling and transfer of user data to be octet oriented, i.e. divisible by eight. According to the invention in the fill bits of a radio block, i.e. in a certain amount of bits reserved for the transfer of user data it is transferred fill bits, depending on channel coding method CS-**1** . . . CS-**4**. This is done by giving the concerned bits a predetermined value, either logical "1" or logical "0". However, in order to get as much user data as possible transferred in a radio block, it is preferably transferred only such a quantity of fill bits, less than an octet, which leaves as many as possible octets available for the transfer of user data. Such quantities are presented in Table 3.

Thence 0 bits (i.e. none) is set as fill bits in channel coding method CS-**1**. In channel coding method CS-**2** for example the seven last bits are chosen as the fill bits, in channel coding method CS-**3** 3 additional bits (for example 3 last bits) are chosen as the fill bits and in channel coding method CS-**4** it is selected 7 additional bits (for example the last 7 bits) as the fill bits. FIG. **4** presents an example of the contents of a radio block payload in the method according to the invention when coding method CS-**2** is used. The payload comprises check bits CHB connected with the performing of the transfer, in which bits it is transferred the MAC-header bits, RLC-header bits, BCS-bits and the tail bits, and transfer bits TB used for the transfer of user data, in which bits it is here transferred the RLC-user data bits and the seven fill bits at the end, which bits otherwise could be used for the transfer of user data. In a radio block according to FIG. **4** it is transferred user data in octets (bytes), in which case handling between different hierarchy levels is kept simple. According to the invention the maximum amount of user data bits transferred in a radio block is obtained by dividing the number of transfer bits TB by eight and by transferring user data in the number of octets (bytes) corresponding to the quotient and by transferring fill bits in the number of transfer bits corresponding to the remainder.

By utilizing the invention it is obtained in each channel coding method an octet oriented number of user data bits or RLC-data bits. At the same it is further obtained after the channel coding and the puncturing presented in Table 1, in each method the desired number of bits, 456. In this way the puncturing need not be changed. This is achieved because the size of the payload is in the method according to the invention kept unchanged with respect to the payloads defined in GSM standard specification 03.64.

Alternatively the payloads of methods CS-**2** and CS-**3** are increased, for example in CS-**2** by one bit to 295 and in CS-**3** by five bits to 343, in which case it would be obtained one more octet for the transfer of user data (with the additional bits noted in Table 3 regarded). Then the number of bits after the convolutional coding would be 590 and 686, in which case the puncturing should be altered by puncturing 134 and 230 bits respectively. If correspondingly the payload should be reduced by 7 and 3 bits, the puncturing should be reduced. Such alternative methods would however require the changing of both the payload and the puncturing in the GSM standard specification 03.64, which is not desirable.

Thanks to the invention data stream through different layers from the higher layers down to the lowest physical layer is made octet oriented, which simplifies the executing of protocols between mobile station MS and fixed network BSS, SGSN. At the same a certain number of bits is lost (0, 7, 3, 7), which bits otherwise could be used for the transfer of user data. When the certain number of bits according to the invention is chosen in such a way that the number is less than an octet and at the same the number of bits in a RLC-data block is adjusted to divisible by eight, the achieving of simpler protocols is however more important than the loss of a few bits for the transfer of user data.

The above has been an introduction of the realization of the invention and its embodiments using examples. It is self evident to persons skilled in the art that the invention is not limited to the details of the above presented examples and that the invention can be realized also in other embodiments without deviating from the characteristics of the invention. The presented embodiments should be regarded as illustrating but not limiting. Thus the possibilities to realize and use the invention are limited only by the enclosed claims. Thus different embodiments of the invention specified by the claims, also equivalent embodiments, are included in the scope of the invention.

What is claimed is:

1. A method for data transfer in a digital mobile communications system, in which method

user data is handled in layers according to protocols,

in one layer of said layers user data is transferred over a physical radio channel between a mobile station and a fixed mobile communications network in radio blocks,

for the transfer of said one layer a payload of a size comprising check bits connected with the performing of the transfer and transfer bits available for the transfer of the user data is formed in the radio block,

each radio block is channel coded using a coding method and the size of said payload is dependent on the coding method, wherein

in the transfer bits of a radio block to be coded using at least said coding method, user data is transferred in a first part of the transfer bits and fill bits are transferred in a second part so that, for the transfer of user data, such a number of transfer bits is chosen which is divisible by eight.

2. A method according to claim **1**, wherein

as the first part of transfer bits for use for the transfer of user data it is chosen the number of octets indicated by the quotient when the number of transfer bits is divided by eight, and

as the second part of transfer bits for use for the transfer of the fill bits it is chosen the number of transfer bits indicated by the remainder of said division.

3. A method according to claim **1**, wherein the radio block is one of the radio blocks according to the GSM standard specification 03.64, except for said fill bits.

4. A method according to claim **1**, wherein in said second part of transfer bits fill bits are set prior to the channel coding performed in the channel coding and prior to interleaving of the bits of the radio block into bursts to be transmitted.

5. A transmitter/receiver device for transmitting user data in a digital mobile communications system, which device comprises

user data handling means for the handling of user data in layers according to protocols,

transmitting means for transmitting user data in radio blocks over a physical radio channel in one layer of said layers,

payload forming means for the forming of a payload of a predetermined size in a radio block for the transfer of said one layer, said payload comprising check bits connected with the performing of the transfer and transfer bits available for the transfer of user data,

channel coding means for the channel coding of a radio block using a coding method, and said size of the payload is dependent on the coding method used, wherein the device comprises

control means for transferring user data in a first part of radio block transfer bits, the radio block being coded using at least said coding method, and for transferring fill bits in a second part of said transfer bits, and said first part of the transfer bits comprises a number of bits divisible by eight.

6. A transmitter/receiver device according to claim **5**, wherein said control means have been arranged to choose as the first part of transfer bits to be used for the transfer of user data the number of octets obtained by the quotient when the number of transfer bits is divided by eight, and

said control means have been arranged to choose as the second part of transfer bits to be used for the transfer of the fill bits the number of bits indicated by the remainder of said division.

7. A transmitter/receiver device according to claim **5**, wherein it has been arranged to transfer transfer bits according to the GSM standard specification 03.64, except for said fill bits.

8. A digital mobile communications system comprising

at least one mobile station and a fixed mobile communications network, which system comprises means for transferring user data over a physical radio channel between the mobile station and the fixed mobile communications network,

user data handling means for the handling of user data in layers according to protocols,

data transfer means for the transfer of user data in radio blocks over a physical radio channel in one layer of said layers,

payload forming means for the forming of a payload of a size in a radio block for the transfer of said one layer, said payload comprising check bits connected with the performing of the transfer and transfer bits available for the transfer of user data,

channel coding means for the channel coding of the radio block using a coding method and the size of said payload is dependent on the coding method,

wherein the system comprises

control means for transferring user data in a first part of transfer bits of the radio block to be coded using at least said coding method, and for the transfer of fill bits in a second part of said transfer bits, and said first part of transfer bits comprises a number of bits divisible by eight.

9. A digital mobile communications system according to claim **8**, wherein

said control means have been arranged to choose as the first part of transfer bits to be used for the transfer of user data the number of octets according to the quotient obtained when the number of transfer bits is divided by eight, and

said control means have been arranged to choose as the second part of transfer bits to be used for the transfer of fill bits the number of transfer bits according to the remainder of said division.

10. A mobile communications system according to claim **8**, wherein it has been arranged to transfer transfer bits according to the GSM standard specification 03.64, except for said fill bits.

11. A method for transceiving a first user data in a digital mobile communications system having a first mobile station and a first fixed mobile communications system, the method comprising the steps of:

selecting a first protocol coding format;

translating the first user data in a first layer according to the first protocol format;

transceiving the first user data over a physical communications channel between the first mobile station and the first fixed mobile communications system, wherein transceiving the first user data comprises the steps of:

selecting a first channel code;

encoding the first user data according to the first channel code, wherein the step of encoding the first user data further comprises the steps of:

using dummy bits where necessary to make the first user data divisible by eight;

encoding the now divisible by eight first user data; and

transceiving the first user data.

12. A method according to claim **11**, wherein the step of encoding the first user data to be divisible by eight further comprises the steps of:

determining a first number of octets of the first user data from the quotient of dividing the first user data by eight; and

determining a second number of dummy bits from the remainder of dividing the first user data by eight.

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EXHIBIT F

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[45] **Date of Patent: Jan. 19, 1999**

[54] **METHOD AND APPARATUS FOR SPEECH TRANSMISSION IN A MOBILE COMMUNICATIONS SYSTEM**

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[51] **Int. Cl.**^{6} **H04B 1/66**

[52] **U.S. Cl.** **375/240**; 375/262; 704/201; 704/501

[58] **Field of Search** 375/240, 262, 375/341, 265; 371/37.4; 704/201, 227, 228, 500, 501

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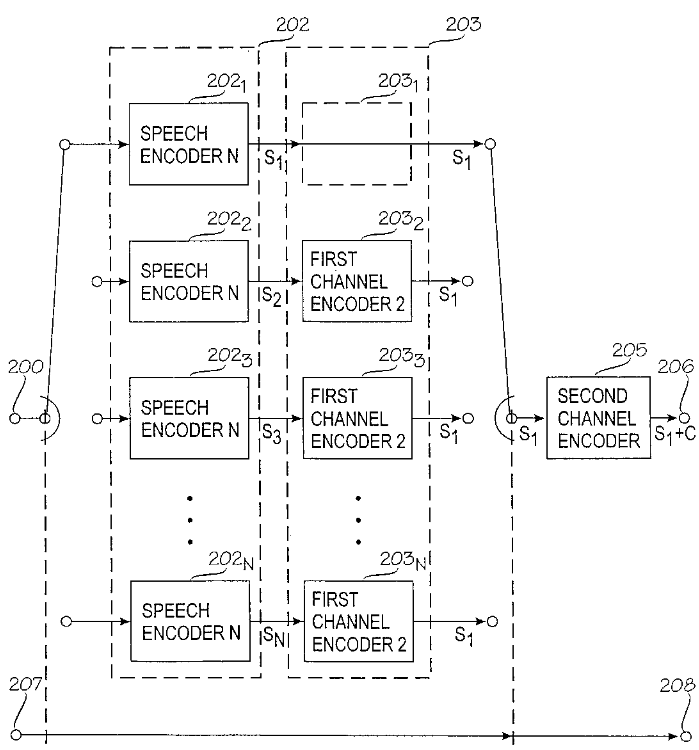
Attorney, Agent, or Firm—Pillsbury Madison & Sutro LLP

[57] **ABSTRACT**

A method and apparatus for speech transmission in a telecommunications system in which a speech signal is compressed to a small number of speech coding bits by a speech coding method, and the speech coding bits are subjected to channel coding. Several different speech coding methods, which may all operate at different transmission rates, are involved in the speech transmission. The method is based on the use of two-stage channel coding. The first channel coding is dependent on the speech coding method, and it is performed in connection with the speech coding in such a manner that the total transmission rate provided by the speech coding and the first channel coding is always constant irrespective of the speech coding method. The second channel coding performed thereafter is always exactly the same regardless of the speech coding method and the first channel coding method, and it is used with all speech coding methods. The second channel coding may be, for example, the original channel coding in an existing telecommunications system.

14 Claims, 2 Drawing Sheets



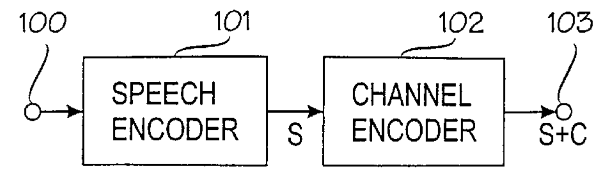


FIG. 1A

PRIOR ART

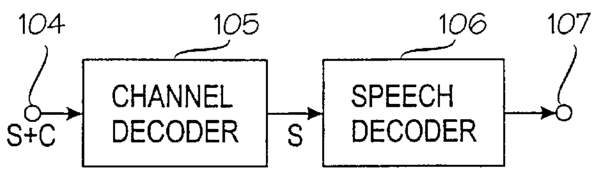


FIG. 1B

PRIOR ART

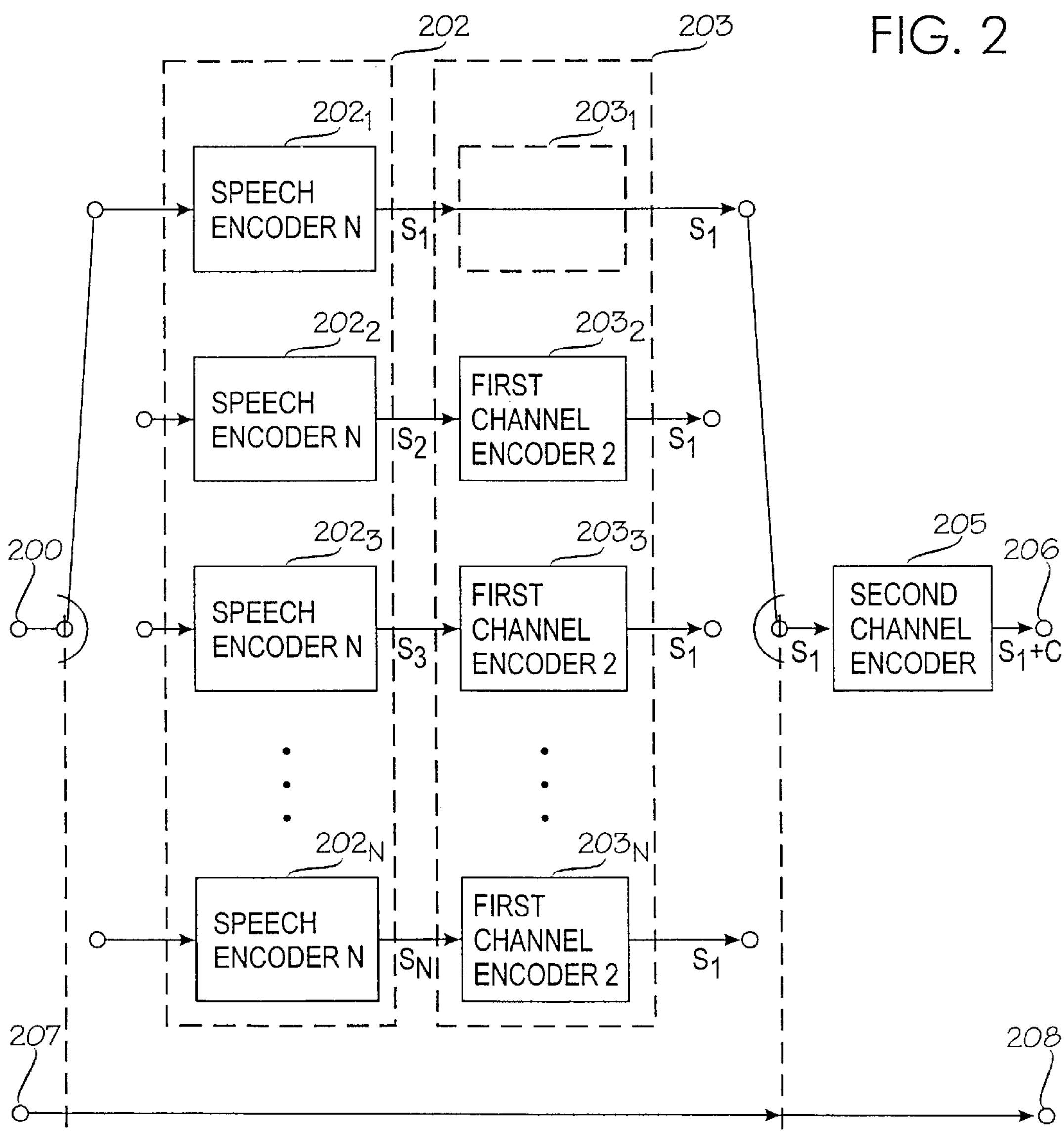
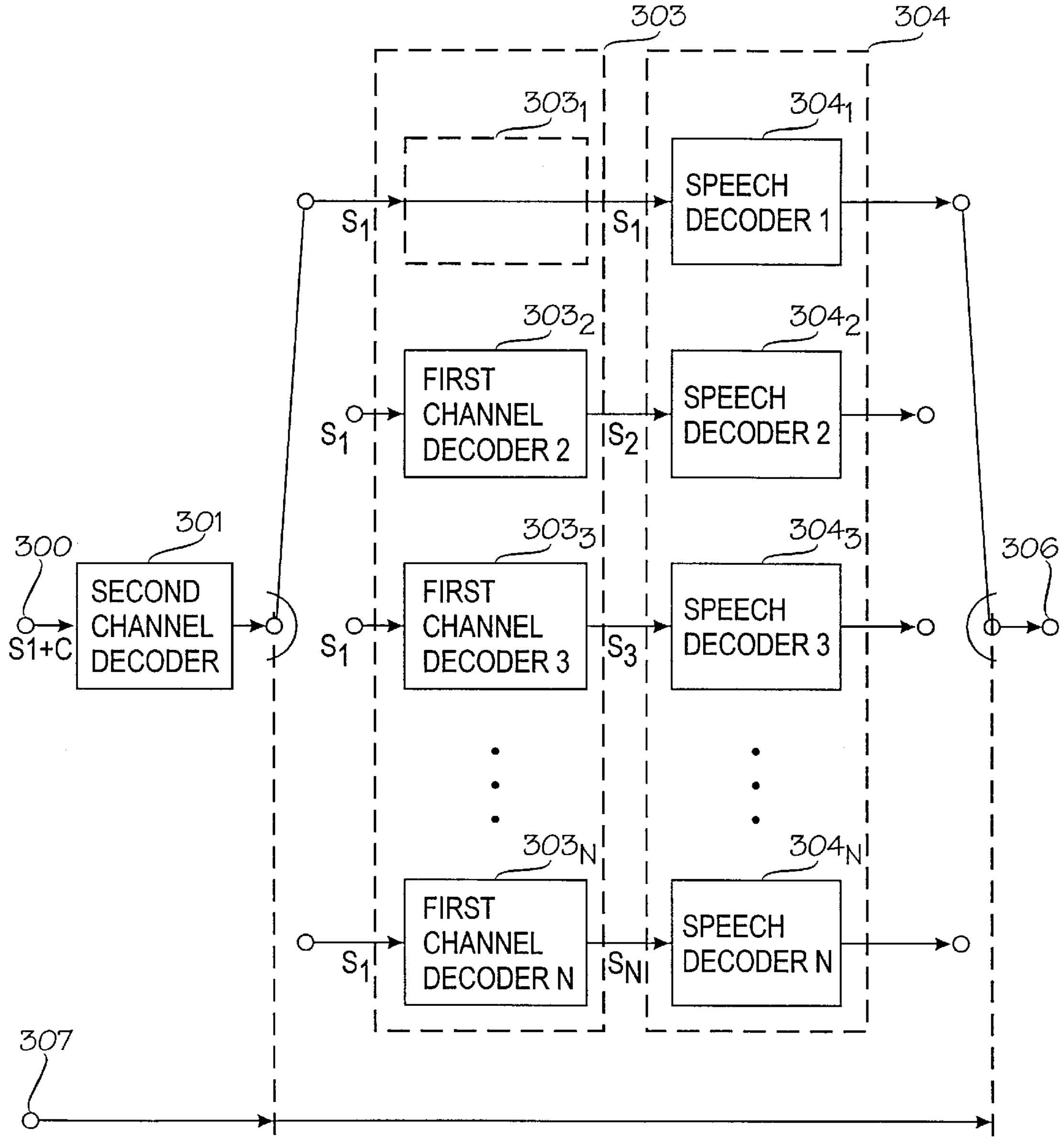


FIG. 2

FIG. 3


303
304
303_1
304_1
SPEECH DECODER 1
S1
S1
303_2
304_2
FIRST CHANNEL DECODER 2
SPEECH DECODER 2
S1
S2
301
300
SECOND CHANNEL DECODER
S1+C
303_3
304_3
FIRST CHANNEL DECODER 3
SPEECH DECODER 3
S1
S3
306
303_N
304_N
FIRST CHANNEL DECODER N
SPEECH DECODER N
S1
SN
307

METHOD AND APPARATUS FOR SPEECH TRANSMISSION IN A MOBILE COMMUNICATIONS SYSTEM

This application claims benefit of international application PCT/ Fl 95/00390 filed Jul. 5, 1995.

FIELD OF THE INVENTION

The invention relates to a method for speech transmission in a mobile communications system, the method comprising compressing a speech signal to a small number of speech coding bits by a speech coding method, and channel encoding the speech coding bits.

BACKGROUND OF THE INVENTION

In telecommunications systems transmitting digital speech, a speech signal is usually subjected to two coding operations: speech coding and channel coding.

Speech coding comprises speech encoding performed in the transmitter by a speech encoder and speech decoding performed in the receiver by a speech decoder. The speech encoder in the transmitter compresses a speech signal so that the number of bits used for representing the speech signal per a unit of time is reduced, whereby less transmission capacity is required for transmitting the speech signal. The speech decoder in the receiver performs a reverse operation and synthesizes the speech signal from the bits generated by the speech encoder. However, the speech synthesized in the receiver is not identical with the original speech compressed by the speech encoder; the original speech has changed more or less as a result of the speech coding. In general, the more the speech is compressed in the speech coding, the more its quality deteriorates during the coding. In the pan-European GSM mobile communication system (Global System for Mobile Communication), for example, the speech encoder of a full-rate traffic channel compresses a speech signal to a transmission rate of 13 kbit/s. The speech synthesized by the corresponding speech decoder is clearly of a poorer quality than the speech transmitted by, for instance, a public switched telephone network (PSTN).

Thus, when a speech coding method is selected, a compromise must be made between the quality offered by the method and the transmission capacity required by it. Another factor to be considered in the selection is the complexity of the implementation of the speech coding method: the quality of speech can usually be improved without increasing the transmission rate if higher requirements for the method as regards calculation capacity and thereby also higher costs of the implementation are allowed. On account of the continuous development of speech coding methods and the implementation techniques, more and more advanced methods are available for speech transmission in the existing telecommunications systems. After the development of the method employed in the GSM, speech coding technology has advanced to such an extent that, as compared with the above-mentioned 13 kbit/s speech coding method, a higher quality of speech can now be achieved at a much lower transmission rate, e.g. 8 kbit/s.

Channel coding comprises channel encoding performed in the transmitter by a channel encoder, and channel decoding performed in the receiver by a channel decoder. The purpose of channel coding is to protect speech coding bits to be transmitted against errors occurring in the transmission channel. Channel coding can either be used for merely detecting whether the transmission has caused any errors in the speech coding bits without any possibility of correcting

them, or it may be capable of correcting errors caused by the transmission, provided that the number of errors does not exceed a given maximum, which is dependent on the channel coding method.

The selection of the channel coding method employed depends on the quality of the transmission channel. In fixed transmission networks the probability of errors is often very low, wherefore not much channel coding is required, whereas in wireless networks such as mobile telephone networks the probability of errors in the transmission channels is often very high, and the channel coding method employed has a significant effect on the resulting quality of speech. Mobile telephone networks usually employ both error-detecting and error-correcting channel coding methods concurrently.

Channel coding is based on the use of error check bits, also called channel coding bits, added to the speech encoding bits. Bits produced by the speech encoder of the transmitter are supplied to a channel encoder, which adds a number of error check bits to them. In the above-mentioned GSM full-rate transmission channel, for example, error check bits with a transmission rate of 9.8 kbit/s are added to speech coding bits of 13 kbit/s on the transmission channel, whereby the total transmission rate of the speech signal on the channel will be 22.8 kbit/s. The channel decoder decodes the channel encoding in the receiver in such a way that only the 13 kbit/s bit stream produced by the speech encoder is applied to the speech decoder. During channel decoding, the channel decoder detects and/or corrects errors that have occurred on the channel as far as such error correction is possible.

Speech coding and channel coding are closely connected with each other in telecommunications systems transmitting speech. The significance of the bits produced by the speech encoder for the quality of speech generally varies so that in some cases one error in an important bit may cause audible noise in the synthesized voice, whereas a larger number of errors in less important bits may be almost imperceptible. How big the differences between the importance of speech coding bits are depends essentially on the speech coding method employed; however, at least small differences can be found in most methods. When a speech transmission method is developed for a telecommunications system, channel coding is therefore usually designed together with speech coding in such a manner that the bits that are the most important for the quality of speech are protected more carefully than less important bits. On a full-rate channel of the GSM system, for instance, the bits produced by the speech encoder are divided into three different categories according to their importance. The most important category is protected in channel coding with both an error-detecting and an error-correcting code; the second most important category is protected only with an error-detecting code; the least important category is not protected at all in channel coding.

Although the speech coding and channel coding are closely connected, there are often considerable differences in their implementation in digital mobile telephone networks. The GSM system may once again be used as an example. Speech encoding and speech decoding are typically carried out by means of software, using a digital signal processor. This applies both to terminal equipment (telephones) and to network elements. Channel coding may also be performed by means of software, but often a separate integrated circuit is designed for this purpose, especially at the network end. Thus, changing of the speech coding method requires often merely a new signal processing program, whereas changing of the channel coding method may require equipment changes.

In addition to the way they are implemented, these two codings, speech coding and channel coding, may differ in their physical locations at the network end of a mobile telephone system. In the GSM system, for example, channel coding in the network is performed in a base station, while speech coding is performed in a separate transcoder unit, which may be remote from the base station, and even if it is located at the base station, it is a completely separate unit. Because of the separate locations, any changes in the transmission rates of the channel coding and speech coding will also entail changes in the connections between the different network elements.

In view of the different ways in which speech and channel coding are performed and their separate locations, it would be clearly more advantageous if the quality of speech could be improved in an existing system merely by changing the speech coding. As the channel coding is, however, usually designed particularly for the speech coding of the existing system, and as the new speech coding method should use exactly the same transmission rate as the original speech coding method of the system, methods for adapting new speech coding methods for existing telecommunications systems have not been disclosed previously.

FIG. **1A** and **1B** are block diagrams illustrating a transmitter and a receiver of a prior art telecommunications system. In the transmitter shown in FIG. **1A**, a speech signal **100** is supplied to a speech encoder **101**, which on the basis of the signal generates compressed speech coding bits having a transmission rate of S kbit/s. These speech coding bits are supplied to a channel encoder **102**, where error check bits are added to them, which results in a total transmission rate of S+C kbit/s. This bit stream **103** is transmitted over the transmission channel to the receiver shown in FIG. **1B**. In the receiver of FIG. **1B**, the bit stream **104** received from the transmitter is at first supplied to a channel decoder **105**, which decodes the channel encoding and transmits the speech coding bits thus obtained to a speech decoder **106**; the transmission rate of the speech coding bits is again S kbit/s. The speech decoder synthesizes a digital speech signal **107**. The telecommunications systems of the prior art thus employ only one speech encoding method and a corresponding channel coding method. Such telecommunications systems include, for example, all the commonest digital mobile telephone systems.

The prior art systems also include systems in which two different speech coding methods are used in such a manner that a separate channel coding method corresponds to each speech coding method, and in which the total transmission rate obtained as a result of the speech and channel coding is different in these two methods. An example of such a system is the GSM mobile telephone system, in which full-rate and half-rate traffic channels are specified.

There are also known solutions in which transmitters and receivers according to FIG. **1A** and **1B** are connected in parallel so that the system that is formed comprises several different speech encoding methods, each of which has a corresponding channel coding method. The speech coding methods used in such a system can operate at different transmission rates, wherefore the channel coding methods corresponding to them are also mutually independent and operate at different transmission rates.

SUMMARY OF THE INVENTION

The object of the present invention is to provide a digital telecommunications system in which several different speech coding methods operating at different transmission rates are used for transmitting speech.

An object of the invention is a method and apparatus which allow the transmission of speech in a digital telecommunications system by the use of several different speech coding methods operating at different transmission rates.

Another object of the invention is a method for adapting more advanced speech coding methods operating at lower transmission rates for an existing digital telecommunications system using a certain speech coding method.

Still another object of the invention is a method for allowing addition of new speech coding methods to a digital telecommunications system without changing the channel coding method originally used.

Yet a further object of the invention is a method for allowing addition of new speech coding methods to an existing digital telecommunications system in such a manner that the addition causes as small changes as possible in the telecommunications system.

This is achieved with a method of the type described in the foregoing BACKGROUND section, which according to the invention is characterized by

using in the transmission of the speech N different speech coding methods, all of which operate at different transmission rates **S1**, **S2**, . . . , and SN kbit/s, respectively, where $N \geqq 2$ and $S1 \geqq S2 \geqq \ldots \geqq SN$,

employing with each speech coding method a first channel encoding method specific for the respective speech coding method, the first channel encoding method comprising adding error-detecting and error-correcting first channel coding bits to the speech coding bits, and producing a constant transmission rate **S1** independent of the speech coding method employed so that the transmission rate of the first channel coding bits added to the speech coding bits during the first channel encoding is, depending on the speech coding method employed, **0**, **S1**–**S2**, . . . , **S1**–SN kbit/s, respectively,

after the first channel encoding, performing a second channel encoding, in which error-detecting and error-correcting second channel coding bits are added to the signal generated by the first channel encoding, the transmission rate of the second channel coding bits being C kbit/s, whereby after the second channel encoding the total transmission rate is a constant **S1**+C kbit/s irrespective of the selected speech encoding method.

The invention also relates to a transmitter apparatus for a telecommunications system transmitting digital speech, the apparatus comprising

speech encoding means for coding a speech signal by a speech coding method,

channel encoding means for channel-encoding the speech-encoded signal to a signal whose transmission rate is equal to the total transmission rate on the transmission channel. The transmitter apparatus is characterized according to the invention in that

the speech encoding means employ two or more speech coding methods, which provide speech-encoded signals having mutually different transmission rates,

the channel encoding performed by the channel encoding means consists of two steps comprising

first channel encodings which are specific for each speech encoding method and which, from the encoded speech signals having different transmission rates, generate first channel-encoded signals having the same constant transmission rate which is independent of the speech coding method, and

a second channel encoding which is independent of the speech coding method and which, from a selected first channel-encoded signal, generates a second channel-encoded signal having a constant transmission rate which is independent of the speech coding method and which is the same as said total transmission rate.

The invention further relates to a receiver apparatus in a telecommunications system transmitting digital speech, comprising

channel decoding means for decoding a received channel-encoded speech signal,

speech decoding means for speech-decoding a channel-decoded speech signal by a speech coding method. The receiver apparatus is characterized according to the invention in that

the speech decoding means employ two or more speech decoding methods for decoding speech-encoded speech signals produced by two or more speech encoding methods and having mutually different transmission rates,

the channel decoding performed by the channel decoding means consists of two steps comprising

a second channel decoding which is independent of the speech coding method and which, from the received channel-encoded speech signal whose constant transmission rate, which is independent of the speech coding method, is the same as the total transmission rate used in the telecommunications channel, produces a first signal having a lower constant transmission rate which is independent of the speech coding method,

first channel decodings which are specific for each speech coding method and which channel-decode the first signal, producing encoded speech signals which are specific for each speech coding method and which have mutually different transmission rates.

According to the invention, speech transmission in a digital telecommunications system employs several different speech coding methods, which may all operate at different transmission rates in such a manner that the total transmission rate obtained as a result of speech coding and channel coding remains the same irrespective of the transmission rate of the speech coding method employed. The method is based on the use of two-part channel coding. The first channel coding is dependent on the speech encoding method and is performed in connection with speech coding in such a way that the total transmission rate provided by the speech encoding and the first channel encoding is always constant irrespective of the speech coding method used. The second channel encoding, subsequently performed, is always exactly the same regardless of the speech encoding method and the first channel encoding method, and it is used with all speech encoding methods. The second channel coding may be, for example, the channel coding originally used in an existing telecommunications system, e.g. channel coding according to the recommendations of the GSM system. In this case, the first channel coding is not used in connection with the speech coding method originally employed in the telecommunications system; in other words, the transmission rate of the first channel coding bits, provided by the first channel encoding, is 0. The first channel coding methods used in connection with speech encoding methods that have been added later and operate at a lower rate provide the same total transmission rate as the original speech encoding method. The new speech encoding methods can thus be added to an existing telecommunications system without changing the channel coding method originally employed. The invention thus allows the quality of voice in an existing system to be improved with as small changes as possible.

The invention differs essentially from, e.g., the GSM system, in which full-rate and half-rate channels are specified, since the invention allows the use of several speech encoding methods in a telecommunications system in such a manner that the total transmission rate used by the speech and channel coding is constant irrespective of the speech encoding method employed. As regards the present invention, the known full-rate and half-rate transmission channels form separate systems, and the invention can be implemented independently in both transmission channels.

The speech coding method employed on each connection and the first channel coding method associated with it can be selected in many different ways, e.g. manually by the user, automatically on the basis of the erroneousness of the transmission path or on the basis of signalling between the transmitter and the receiver.

The method of the invention is thus based on the use of first channel coding in such a manner that a constant transmission rate is obtained as a result of speech encoding and the first channel encoding; a new speech coding method can be adapted for an existing system without changing the originally used second channel coding method. It is particularly characteristic of the invention that, when it is applied to an existing system, all speech coding methods have a common second channel encoder, whereas a separate first channel coding method is associated with each speech coding method in such a way that one speech encoder is not provided with any kind of first channel encoder.

The invention can be implemented in such a way that the first channel encoding is performed in connection with speech encoding, whereby the originally used channel coding unit, which performs the second channel coding, can be retained unchanged. In the GSM mobile telephone network, for example, the first channel coding can be carried out by means of software in a transcoder unit together with the new speech coding method, in which case no other changes are required at the fixed network end. In a terminal equipment (telephone), the new speech coding method and the corresponding first channel coding method can be implemented using the signal processor of the telephone in the same way as in the originally used speech coding.

BRIEF DESCRIPTION OF THE DRAWINGS

In the following, the invention will be described in greater detail by means of preferred embodiments with reference to the accompanying drawings, in which:

FIG. **1**A and FIG. **1**B illustrate a transmitter and a receiver of the prior art, respectively,

FIG. **2** is a block diagram of a transmitter in a telecommunications system according to the invention,

FIG. **3** is block diagram of a receiver in a telecommunications system according to the invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS OF THE INVENTION

The present invention is particularly suitable for telecommunications systems in which channel coding is particularly significant. The main field of application of the invention is wireless speech transmission, e.g. in digital mobile telephone systems. A particularly important field of application for the invention is the GSM mobile telephone system and its derivatives which are similar to the GSM as regards speech coding and channel coding but which may differ from the GSM for instance in their operating frequency ranges, such as the DCS-1800 and DCS-1900 systems.

As stated above, speech transmission in a digital telecommunications system of the invention employs several different speech coding methods which do not all operate at the same transmission rate. A first channel coding method is assigned to each speech coding method in such a way that the total transmission rate obtained as a result of speech coding and channel coding is kept constant regardless of the transmission rate of the speech coding method employed. The second part of channel coding is always exactly the same irrespective of which speech coding method and first channel coding method are used.

FIG. **2** is a block diagram of a transmitter in a telecommunications system according to the invention. The transmitter comprises N parallel speech encoders **202**$_1$ –**202**$_N$; the transmission rates of the compressed speech signals generated by these encoders, i.e. of the speech coding bits, are S_1, S_2. . . , S_N kbit/s, respectively (for reasons of clarity, the unit kbit/s for the transmission rate will be omitted below). A digital speech signal **200** to be transmitted is supplied to an input switch **201**, which is used to select one of these N speech encoders **202** for each speech connection. In the embodiment illustrated in FIG. **2**, the invention is applied in such a way that new, more advanced speech coding methods are added to an existing system. Therefore the speech encoder **202**$_1$ in Figure **1**A employs a speech coding method that has originally been used in the existing telecommunications system and that provides a transmission rate of S_1 for the speech coding bits, which is the same as the transmission rate of speech coding bits used originally in the existing telecommunications system. In the transmitter, it is thus also possible to select N−1 other speech encoders **202**$_2$–**202**$_N$, which provide transmission rates of S_2, S_3 ,. . . S_N, respectively, where the total number of speech encoders is $N \geqq 2$. The transmission rates of the speech encoders have the following relationship: $S_1 \geqq S_2 \geqq S_3 \geqq . . . \geqq S_N$, where $S_1 \geqq S_N$ must be true. The transmission rates of speech coding bits used by the speech encoders added to the telecommunications system may thus be, for some of the speech encoders **202**$_2$–**202**$_N$ the same as the transmission rate S_1, originally used for speech coding bits in the telecommunications system; but at least for one speech encoder this transmission rate is lower than the transmission rate S_1 originally used. Each speech encoder **202**$_1$–**202**$_N$ is used with a channel encoder **203**$_1$–**203**$_N$ specific for the respective speech coding method; however, in the case of those speech encoders which provide a transmission rate of S_1 for the speech coding bits, the first channel encoder **203** does not affect the speech coding bits in any way but forwards them as such to the second channel encoder **205**. In this case, the transmission rate provided by the first channel encoder for the channel coding bits, is thus 0, and the transmission rate to the second channel encoder **205** is S_1. In other words, the first channel encoder **203** is, in fact, omitted from this embodiment, like the first speech encoder **202**$_1$ in FIG. **2**, corresponding to the speech coder **202** originally used. A first channel coding bit rate equal to that of the first channel encoder **203**$_2$–**203**$_N$ may also be provided by another speech encoder **202**$_2$–**202**$_N$, if the speech coding bit rate of the respective speech encoder is S_1. In other cases, the first channel encoder **203**$_2$–**203**$_N$ adds error correction bits to the bit stream generated by the corresponding speech encoder **202**$_2$–**202**$_N$ so that the total transmission rate provided by the speech encoding and the first channel encoding is S_1 irrespective of the speech encoding method used. The channel coding bit rate provided by the first channel encoders **203**$_1$–**203**$_N$ is thus correspondingly **0**, S_1–S_2, S_1–S_3, . . . , S_1–S_N, depending on the speech encoding method employed by the speech encoder **202**$_1$–**202**$_N$ connected in series before them. It is characteristic of the invention that there is at least one speech encoding method whose respective first channel encoder **203** provides a first channel coding bit rate higher than 0. From the first channel encoder **203**$_1$–**203**$_N$ selected for the speech connection, the speech coding bits and the first channel coding bits are supplied via a switch **204** to a second channel encoder **205**. In the transmitter according to the invention, the transmission rate of the bit stream to be transmitted to the second channel encoder **205** is thus a constant S_1 kbit/s. The second channel encoder adds error correction bits to the bit stream produced by the selected speech encoder **202** and the first channel encoder **203** so that, at the output **206** of the second channel encoder, the total transmission rate is a constant S_1+C kbit/s. The switches **201** and **204** are controlled synchronically by a control signal **207** so that they will select the series connection of the speech encoder **202** which implements the desired speech encoding method, and the respective first channel encoder **203**. Information on the selected speech coding method is also sent to the transmission channel in a signal **208** to enable the receiver to select the correct first channel decoding and speech decoding methods corresponding to the speech encoding and first channel encoding methods used.

FIG. **3** is a block diagram of a receiver in a telecommunications system according to the invention. The receiver comprises a second channel decoder **301**, a selection switch **302**, N parallel first channel decoders **303**$_1$–**303**$_N$, N parallel speech decoders **304**$_1$–**304**$_N$, and a selection switch **305**. The receiver receives the speech coding bits and the first and second channel coding bits from the transmitter through the transmission channel at the input **300** of the second channel decoder **301**. The second channel decoder **301** decodes the second channel encoding performed by the second channel encoder **205** of the transmitter shown in FIG. **2**; as a result of this, the transmission rate of the bit stream received at the input **300** decreases from the constant S_1+C to the constant S_1. The second channel decoder **301** is thus independent of the speech coding method used and always performs the same channel decoding. The bit stream from the output of the second channel decoder **301**, having the transmission rate of S_1, is supplied to the selection switch **302**. The selection switch **302** switches the output of the second channel decoder **301** to one of N first channel decoders **303**$_1$–**303**$_N$, depending on the speech coding method used. The receiver also receives a signal **307** from the transmitter, through the transmission channel. Signal **307** corresponds to signal **208** of FIG. **2** and gives information on the speech coding method employed on the speech connection; the state of switch **302** and also that of switch **305** are determined on the basis of this information. The first channel decoder **303** is always dependent on the speech coding method employed, and it is connected in series with the speech decoder **304** assigned to it. The first channel decoder **303** decodes the first channel encoding performed by the first channel encoder **203** of the transmitter shown in FIG. **2** and provides the transmission rate used by the selected speech coding method. If the transmission rate of the channel coding bits added by the first channel encoding is 0, as in the case of channel encoder **203**$_1$ in FIG. **2**, the respective first channel decoder feeds the bits received from the second channel decoder **301**, the transmission rate of said bits being S_1, directly to the associated speech decoder; thus, in fact, there is no first channel decoder for such a speech coding method. In FIG. **3**, first channel decoder $\mathbf{303}_1$—corresponding to the missing first channel encoder $\mathbf{203}_1$ of FIG. **2** —is omitted. Naturally only one of the channel decoders is in use at a time. The other first channel decoders $\mathbf{303}_2$–$\mathbf{303}_N$ of FIG. **3** decode the channel encoding associated with the speech encoders $\mathbf{202}_2$–$\mathbf{202}_N$ of FIG. **2** and thus decrease the constant transmission rate S_1 of the bit stream received from the second channel decoder **301** by the transmission rate **0**, S_1–S_2, S_1–S_3, . . . , S_1–S_N, providing the transmission rates S_1, S_2, . . . , S_N kbit/s, which are dependent on the speech coding methods. The bit stream generated by the first channel decoder $\mathbf{303}_1$–$\mathbf{303}_N$ is supplied to the corresponding speech decoder $\mathbf{304}_1$–$\mathbf{304}_4$, which by means of the received speech coding bits generates a synthesized speech signal. The output signal of the speech decoder of the selected speech coding method is switched by selection switch **305** to the output **306** of the receiver. The position of the selection switch **305** is determined on the basis of signal **307** received from the transmitter through the transmission channel.

In the embodiment of the invention illustrated in FIGS. **2** and **3**, information on the speech coding method employed is forwarded through the transmission channel from the transmitter to the receiver. This information transfer may be based on any suitable method, e.g. a signalling method known per se. The speech coding method may also be permanent at each receiver or transmitter. It is, however, essential that both the transmitter and the receiver have information on the speech coding method employed so that the positions of the switches **201**, **204**, **302** and **305** can be determined correctly, and the same speech coding method can be selected both in the transmitter and in the receiver.

There are several ways of selecting the speech coding method according to the invention for each speech connection. Some factors influencing the selection and a few selection method will be described in the following; however, the invention is not limited to these examples.

If the transmission rate used for transmitting the speech coding bits is as low as possible in the selected speech coding method, more bits are left for the first channel coding. In this case, the performance of the system will be improved in an erroneous channel, but, on the other hand, it may be decreased in an error-free channel, where it would be advantageous to use as much capacity as possible for speech coding. To classify speech coding methods as error-tolerant methods and methods suitable for a high-quality transmission channel according to the transmission rate is a very coarse simplification, because transmission rate is not the only significant factor. In this connection, however, such a classification will clarify the selection of the speech coding method. According to one embodiment of the invention, the speech coding method is selected according to the erroneousness of the transmission channel: in the case of a high quality transmission channel, a speech coding method is selected in which a major part of the transmission channel capacity is used for speech coding, i.e. the speech coding has a high transmission rate; in the case of a poor quality transmission channel, a speech coding method is selected in which the first channel coding is emphasized more, i.e. the speech coding method has a low transmission rate. The selection can be made by monitoring the quality/erroneousness of the transmission channel when the connection is established. Since the quality of a transmission channel may vary to a great extent during a speech connection, the quality/erroneousness can also be monitored during the speech connection, and if necessary, the speech coding method can be changed.

One of the most significant advantages of the present invention is that it allows new speech coding methods to be added to an existing telecommunications system. In such a case, it is essential that the transmitters and receivers using the new speech coding methods still operate together with the transmitters and receivers which have originally been used in the telecommunications system and in which these new speech coding methods have not been implemented. A telecommunications system of this kind typically comprises various transmitters and receivers which do not employ a uniform group of speech coding methods. However, all transmitters and receivers must be able to use at least one speech coding method which is common to all of them. If, for example, the new system of the invention is provided by planning and realizing a new speech coding method for use on a full-rate GSM transmission channel, the new GSM system will comprise telephones and network elements provided with both the new speech coding method and the method currently in use in the GSM system (i.e. new equipments). In addition, the system will also comprise equipments which have already been used previously and which are provided only with the current speech coding method according to the GSM system (i.e. old equipments). When a new and an old equipment communicate with each other, the speech coding method selected must be the current GSM method, which can be selected in both equipments. Another factor influencing the selection is thus a heterogeneous group of equipments: the speech coding method selected at the beginning of a connection is the most suitable speech coding method which can be selected both in the transmitter and in the receiver. Even in this case, it may be necessary to change the speech coding method during the connection, as the transmitter and receiver may also be changed during one speech connection.

In the description of the invention, it has been stated that the channel encoder **203**, **205** adds error check bits to the speech coding bits. The channel encoder **203**, **205** can, in fact, operate in practice so that the error check bits are added to the speech coding bits generated by the speech encoder in such a manner that the speech coding bits are still visible and unchanged in the resulting bit error which now also includes the channel coding bits. Depending on the channel coding method, the channel encoder **203**, **205** may also operate so that even the speech coding bits are changed when error check bits are added. In this case, the channel decoder **301**, **303** restores the speech coding bits at channel decoding if the number of errors received from the channel does not exceed the error correction capability of the channel coding. In view of the present invention, there is no difference between these two methods, and the expression "the channel encoder adds error check bits to the speech coding bits" refers to both these cases, since, from the point of view of speech coding and the transmission rate, this is exactly what happens.

In a telecommunications system, a speech signal is often also subjected to other operations, such as encryption, interleaving of the bits to be transmitted (closely connected with channel coding), a possible precoding associated with the modulation method, or bit interleaving in association with spectral shaping of the signal; however, these methods are irrelevant to the present invention. In the present invention, channel coding refers particularly to the use of error-detecting and error-correcting codes. Depending on how the channel coding is implemented, the channel decoder may have available data on the results of demodulation so that it may utilize data on the error probabilities of individual bits, i.e. the so-called soft decisions of the demodulator. It is not relevant to the invention whether soft decisions are available or not, and the channel decoder of the invention covers both cases. In typical implementations, the second channel decoder **301** of the invention has the results of soft decisions available, whereas the first channel decoder **303** does not, but the system of the invention may also be implemented in some other way.

The following is a simple general example of how a new speech coding method is added to a full-rate transmission channel of the GSM mobile telephone system for use in conjunction with the RPE-LTP speech coding method presently in use. The example is given merely to illustrate the invention; it is thus only one possible embodiment, and the invention is not limited to it. In an RPE-LTP speech coding method, a speech signal is divided into frames of 20 ms, of each of which an RPE-LTP speech encoder forms 260 speech coding bits, whereby the transmission rate of the speech coding bits is 13 kbit/s. The channel encoder used on a full-rate GSM speech channel, i.e. the second channel encoder in the system of the invention, adds 196 error coding bits to 260 speech coding bits; the total bit number in one 20 ms frame is thus 456 bits, which corresponds to a total transmission rate of 22.8 kbit/s. In this example, a technically highly advanced speech coding method in which the transmission rate of speech coding bits is 8 kbit/s and the speech signal is divided into frames of 10 ms, each containing 80 bits, is added to a full-rate GSM speech channel. To implement the invention, a first channel coding method must be designed for this speech coding method. A simple exemplary solution for a first channel coding method consists of the following operations, which are described from the point of view of the first channel encoder:

(a) Two speech frames of 80 bits are combined in the first channel encoder into one frame of 160 bits.

(b) 100 error-correcting bits are added to these 160 bits by an error-correcting code known per se. The selection of the code is influenced by both the 8 kbit/s speech coding method employed and the channel coding method of the full-rate GSM channel. This results in a 260-bit speech frame according to a full-rate GSM channel.

(c) The 260 bits generated are classified into three groups according to their importance in view of the channel coding of the full-rate GSM channel.

Thereafter the speech coding bits and the first channel coding bits are supplied to the second channel encoder, which is identical with the channel encoder used in connection with the RPE-LTP. In the receiver, the first channel decoder decodes the first channel encoding.

The figures and the description associated with them are intended merely to illustrate the present invention. In their details, the method, receiver and transmitter of the invention may vary within the scope of the appended claims.

We claim:

1. A method for speech transmission of a speech in a telecommunications system, said method comprising the steps of:

compressing a speech signal of the speech to a small number of speech coding bits by speech coding method,

channel encoding the speech coding bits, and

using, in transmitting the speech as a transmitted signal by a transmitter, N different speech coding methods, all of which operate at different transmission rates **S1**, **S2**, . . . , and SN kbit/s, respectively, where N≧2 and **S1** ≧**S2**≧. . . ≧SN, including:

employing with each speech coding method a first channel encoding method specific for the respective said speech coding method, said first channel encoding method comprising adding error-detecting and error-correcting first channel coding bits to the speech coding bits, and producing a constant transmission rate **S1** which is independent of the speech coding method employed, so that the transmission rate of the first channel coding bits added to the speech coding bits during the first channel encoding its, depending on the speech coding method employed, **0** **S1**–**S2**, . . . , **S1**–SN kbit/s, respectively, and

after the first channel encoding, performing a second channel encoding, in which error-detecting and error-correcting second channel coding bits are added to the signal generated by the first channel encoding, the transmission rate of said second channel coding bits being C kbit/s, whereby, after the second channel encoding, the total transmission rate is a constant **S1**+C kbit/s irrespective of the selected speech encoding method.

2. The method according to claim **1**, further including:

receiving the transmitted signal, including:

performing a second channel decoding for removing the second channel coding bits, which were added by the second channel encoding and the transmission rate of which is C kbit/s, and

performing, after the second channel decoding, a first channel decoding for removing the first channel coding bits added by the first channel encoding, in such a manner that, depending on the speech coding method employed, a transmission rate of **S1**, **S2**, . . . , SN kbit/s, respectively, is provided for the speech coding bits to be supplied to the speech decoding.

3. The method according to claim **1** or **2**, further comprising:

classifying the bits supplied to the second channel coding into several groups according to their importance for error protection in the speech coding method employed, in such a manner that the error correction capability of the second channel coding is directed to the bits that are the most important for each speech coding method and the first channel coding method employed in conjunction with the respective speech coding method.

4. The method according to claim **1** or **2**, wherein:

when a speech connection is established, the speech coding method employed is selected according to the erroneousness of the transmission path in such a way that the more there are transmission errors on the transmission channel, the lower is the transmission rate of the speech coding bits to the first channel encoding from the selected speech coding method, and the higher is the transmission rate of the first channel coding bits.

5. The method according to claim **4**, further comprising:

changing the speech coding method during the speech connection when the erroneousness of the transmission path changes in such a way that the more there are transmission errors on the transmission path, the lower is the transmission rate of the speech coding bits supplied to the first channel encoding from the selected speech coding method.

6. The method according to claim **1** or **2**, in an instance wherein:

a corresponding speech decoding method cannot be selected in all receivers of a plurality of receivers for all the speech coding methods that can be selected in the transmitter, and

a corresponding speech coding method cannot be selected in all transmitters of a plurality of transmitters for all the speech decoding methods that can be selected in the different receivers, and

said method further comprising:

when a speech connection is established, the speech coding method employed is selected by means of signalling between the respective transmitter and the respective receiver so that the speech coding method selected is a speech coding method which can be selected both in the respective transmitter and in the respective receiver.

7. The method according to claim **1** or **2**, in an instance wherein:

a corresponding speech decoding method cannot be selected in all receivers of a plurality of receivers for all the speech coding method which can be selected in all transmitters of a plurality of transmitters, and

a corresponding speech coding method cannot be selected in all of the transmitters for all the speech decoding methods which can be selected in all of the receivers, and

said method further comprising:

manually selecting the speech coding method employed before a speech connection is established in such a way that the speech coding method selected is a speech coding method which can be selected both in the respective transmitter and in the respective receiver.

8. The method according to claim **1**, further comprising:

carrying out the first channel encoding in more than one step, and

the second channel encoding being the same in the case of all speech coding methods.

9. A transmitter apparatus for a telecommunications system transmitting digital speech on a transmission channel, said apparatus comprising:

speech means for coding a speech signal by a speech coding method to provide a speech encoded signal,

channel encoding means for channel-encoding the speech-encoded signal to a signal whose transmission rate is equal to a total transmission rate on the transmission channel,

the speech encoding means employing two or more speech coding methods, which provide speech-encoded signals have mutually different transmission rates (**S1**, **S2**, . . . , SN),

the channel encoding means being arranged to provide the channel encoding in two steps comprising

first channel encodings which are specific for each speech encoding method and which, from the encoded speech signals having different transmission rates, generate first channel-encoded signals having a same constant transmission rate (**S1**) which is independent of the speech coding method, and

a second channel encoding which is independent of the speech coding method and which, from a selected

first channel-encoded signal, generates a second channel-encoded signal having a constant transmission rate (**S1**+C) which is independent of the speech coding method and which is the same as said total transmission rate.

10. The transmitter apparatus according to claim **9**, comprising:

means for selecting the speech encoding method and the first channel encoding, and for switching the signal produced by the selected speech encoding method and first channel encoding to the second channel encoding.

11. The method according to claim **10**, wherein:

said means for selecting are arranged to be controlled on the basis of the erroneousness of the transmission path, or on the basis of signalling between the transmitter apparatus and a receiver apparatus, or manually.

12. A receiver apparatus in a telecommunications system transmitting digital speech on a telecommunications channel at a total transmission rate, comprising:

channel decoding means for decoding a received channel-coded speech signal,

speech decoding means for speech-decoding a channel-decoded speech signal by a speech decoding method,

the channel decoding means being arranged to provide channel decoding two steps comprising:

a second channel decoding which is independent of the speech coding method and which, from said received channel-encoded speech signal, sharing a constant transmission rate (**S1**+C), which is independent of the speech coding method and is the same as the total transmission rate used in the telecommunications channel, produces a first signal having a lower constant transmission rate (**S1**) which is independent of the speech coding method, and

first channel decodings which are specific for each speech coding method and which channel-decode said first signal, producing encoded speech signals which are specific for each speech coding method and which have mutually different transmission rates (**S1**, **S2**, . . . , **S3**), and

the speech decoding means employ two or more speech decoding methods for decoding said speech-coded speech signals produced by two or more speech encoding methods and having mutually different transmission rates (**S1**, **S2**, . . . , SN).

13. The receiver apparatus according to claim **12**, comprising:

means for selecting the speech decoding method employed and the first channel decoding specific for the selected speech decoding method, and for switching the signal produced by the second channel decoding (**301**) to the selected speech decoding method and selected first channel decoding method.

14. The receiver apparatus according to claim **13**, wherein:

the selection means are arranged to be controlled on the basis of the erroneousness of the transmission path, or on the basis of signalling between a transmitter apparatus and the receiver apparatus, or manually.

* * * * *

EXHIBIT G

US005946651A

United States Patent [19]

Jarvinen et al.

[11] **Patent Number: 5,946,651**

[45] **Date of Patent: Aug. 31, 1999**

[54] **SPEECH SYNTHESIZER EMPLOYING POST-PROCESSING FOR ENHANCING THE QUALITY OF THE SYNTHESIZED SPEECH**

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[73] Assignee: **Nokia Mobile Phones**, Salo, Finland

[21] Appl. No.: **09/135,936**

[22] Filed: **Aug. 18, 1998**

Related U.S. Application Data

[63] Continuation of application No. 08/662,991, Jun. 13, 1996.

[51] **Int. Cl.**[6] **G10L 9/14**

[52] **U.S. Cl.** **704/223**; 704/208; 704/219; 704/220; 704/222

[58] **Field of Search** 704/200, 207, 704/223, 219, 221, 222, 220, 208

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Primary Examiner—David R. Hudspeth
Assistant Examiner—Vijay B. Chawan
Attorney, Agent, or Firm—Perman & Green, LLP

[57] **ABSTRACT**

A post-processor **317** and method substantially for enhancing synthesised speech is disclosed. The post-processor **317** operates on a signal ex(n) derived from an excitation generator **211** typically comprising a fixed code book **203** and an adaptive code book **204**, the signal ex(n) being formed from the addition of scaled outputs from the fixed code book **203** and adaptive code book **204**. The post-processor operates on ex(n) by adding to it a scaled signal pv(n) derived from the adaptive code book **204**. A gain or scale factor p is determined by the speech coefficients input to the excitation generator **211**. The combined signal ex(n)+pv(n) is normalised by unit **316** and input to an LPC or speech synthesis filter **208**, prior to being input to an audio processing unit **209**.

46 Claims, 7 Drawing Sheets

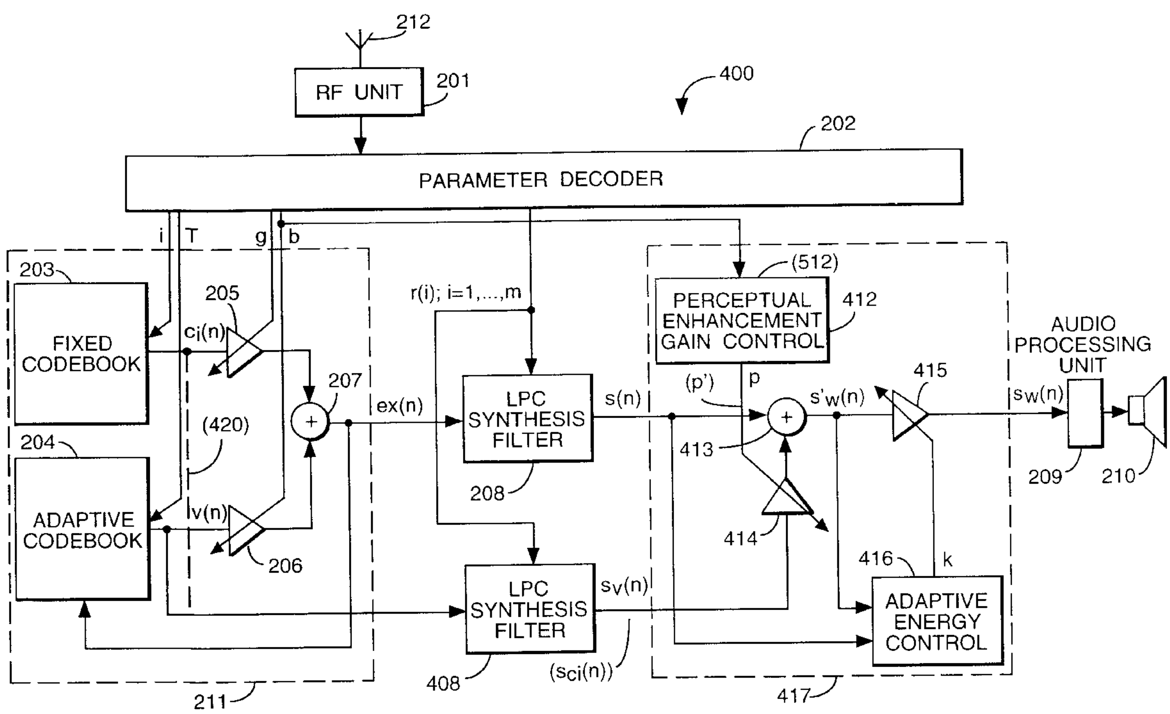


Fig.1.
PRIOR ART

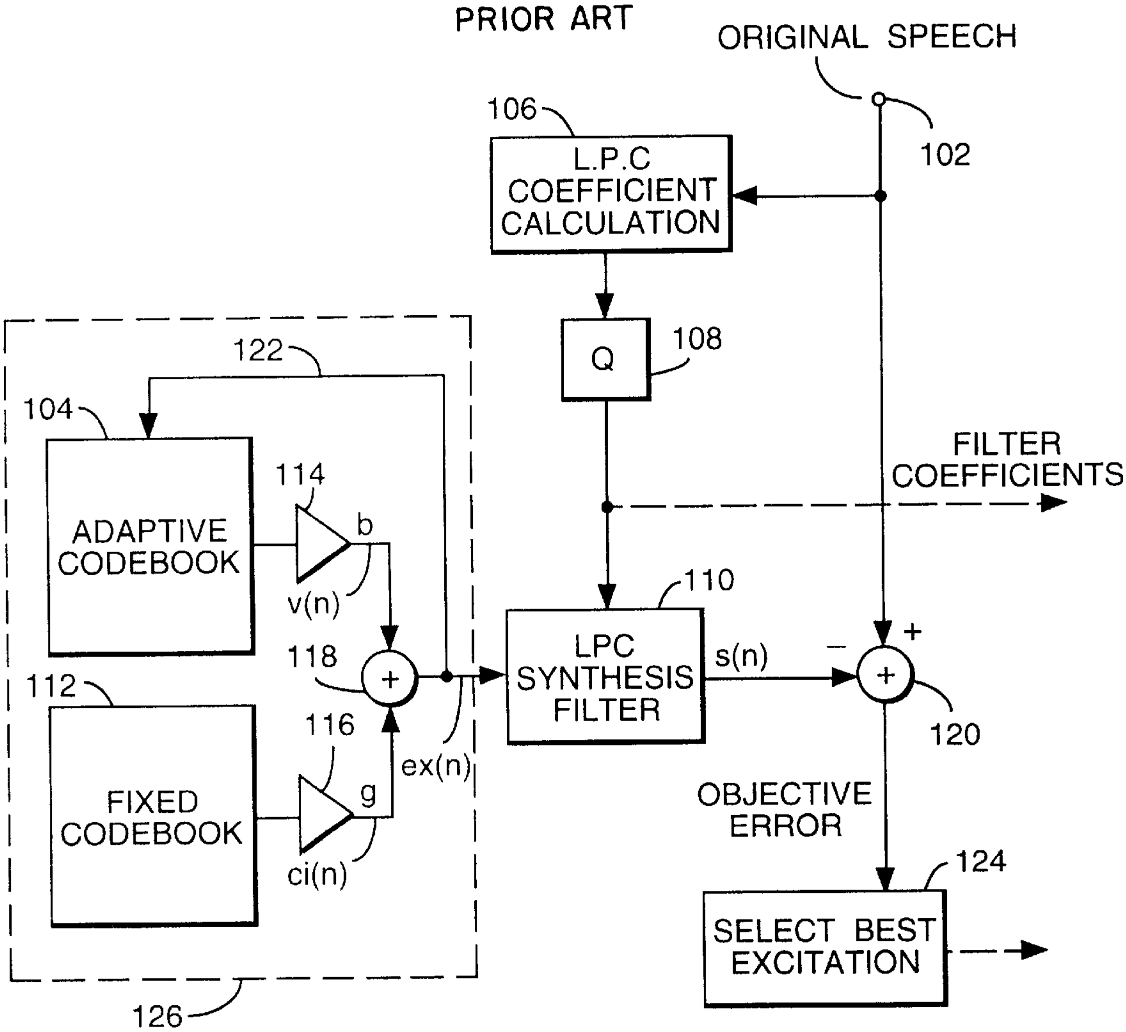
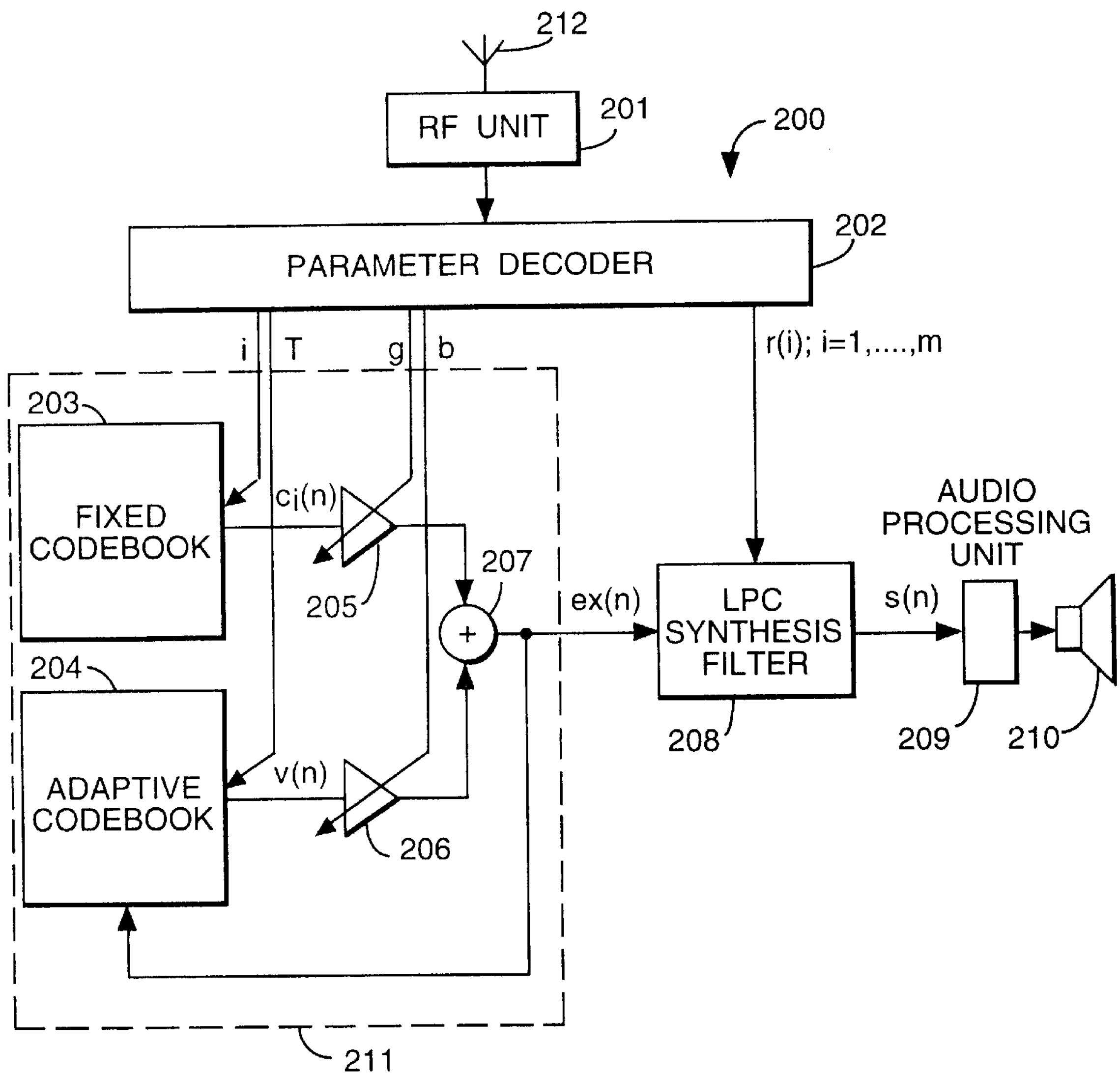


Fig.2. PRIOR ART



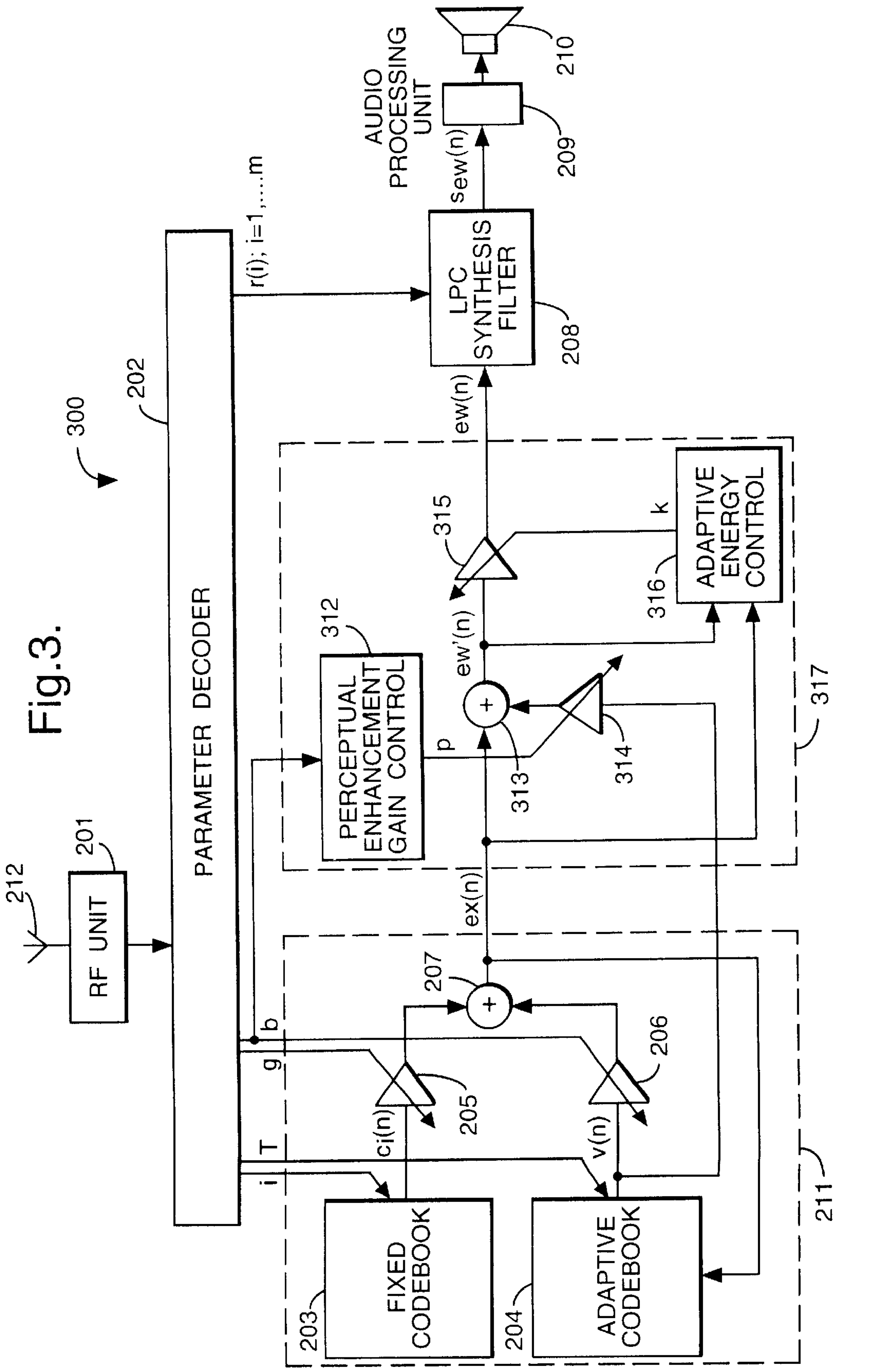

300
212
201
RF UNIT
202
PARAMETER DECODER
r(i); i=1,...,m
i
T
g
b
FIXED CODEBOOK
203
ADAPTIVE CODEBOOK
204
ci(n)
v(n)
205
206
207
211
ex(n)
PERCEPTUAL ENHANCEMENT GAIN CONTROL
312
p
313
314
ew'(n)
315
k
ADAPTIVE ENERGY CONTROL
316
317
ew(n)
LPC SYNTHESIS FILTER
208
sew(n)
AUDIO PROCESSING UNIT
209
210

Fig.3.

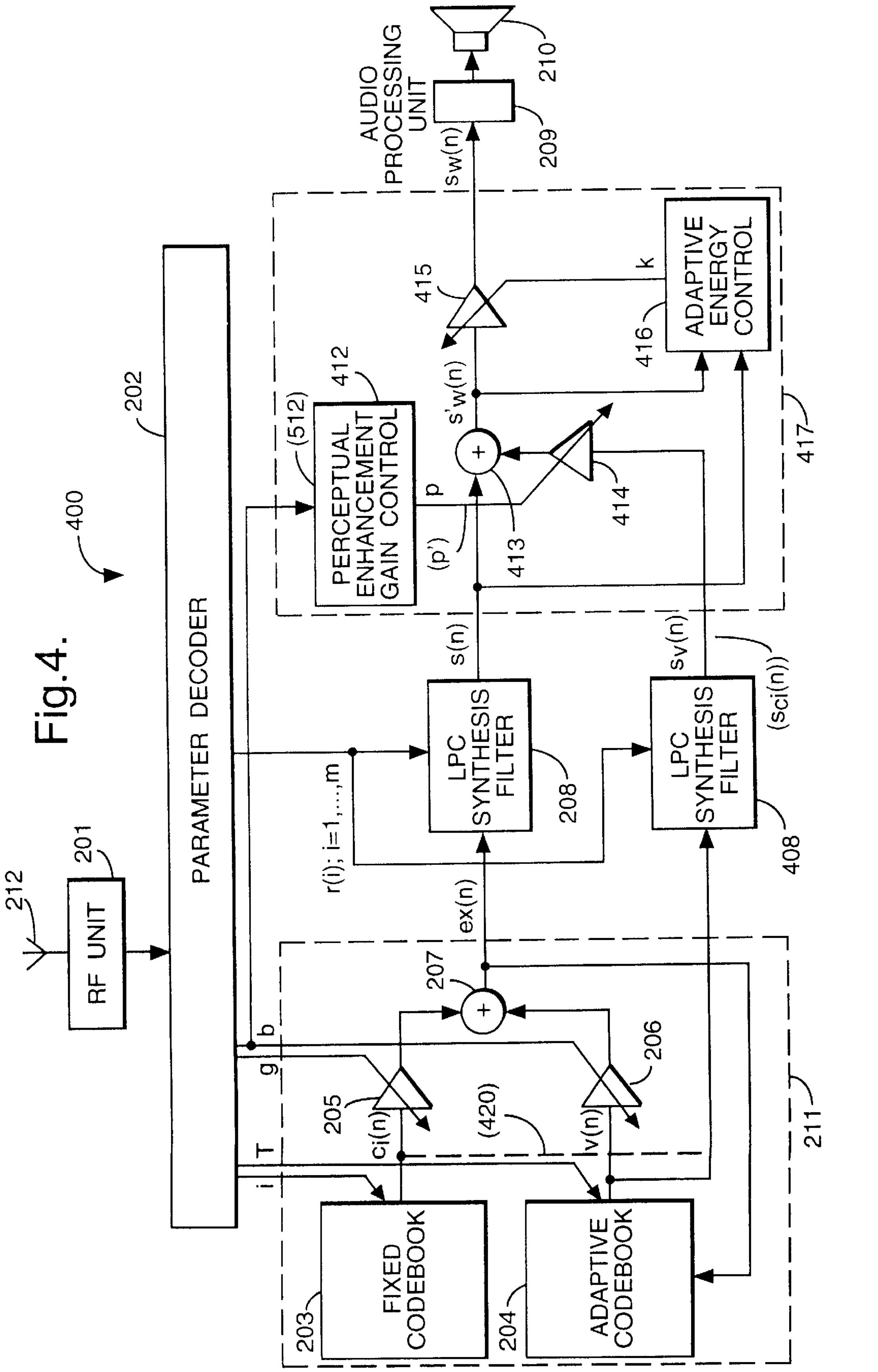
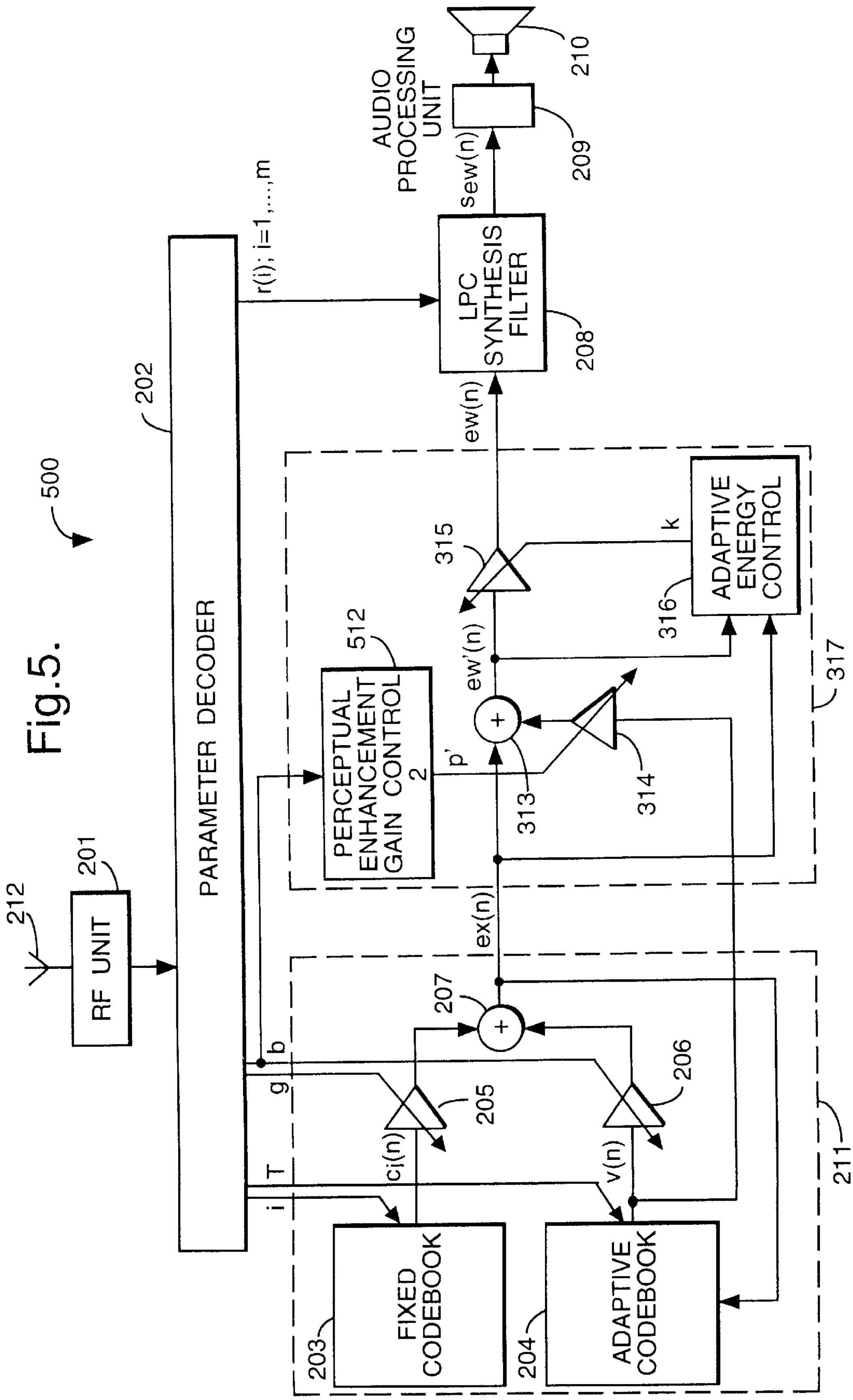

400
212
RF UNIT
201
PARAMETER DECODER
202
FIXED CODEBOOK
203
ADAPTIVE CODEBOOK
204
205
206
207
211
(420)
i
T
g
b
ci(n)
lv(n)
ex(n)
r(i); i=1,...,m
LPC SYNTHESIS FILTER
208
LPC SYNTHESIS FILTER
408
s(n)
sv(n)
(sci(n))
PERCEPTUAL ENHANCEMENT GAIN CONTROL
412
(512)
p
(p')
413
414
s'w(n)
415
k
ADAPTIVE ENERGY CONTROL
416
417
sw(n)
AUDIO PROCESSING UNIT
209
210

Fig.4.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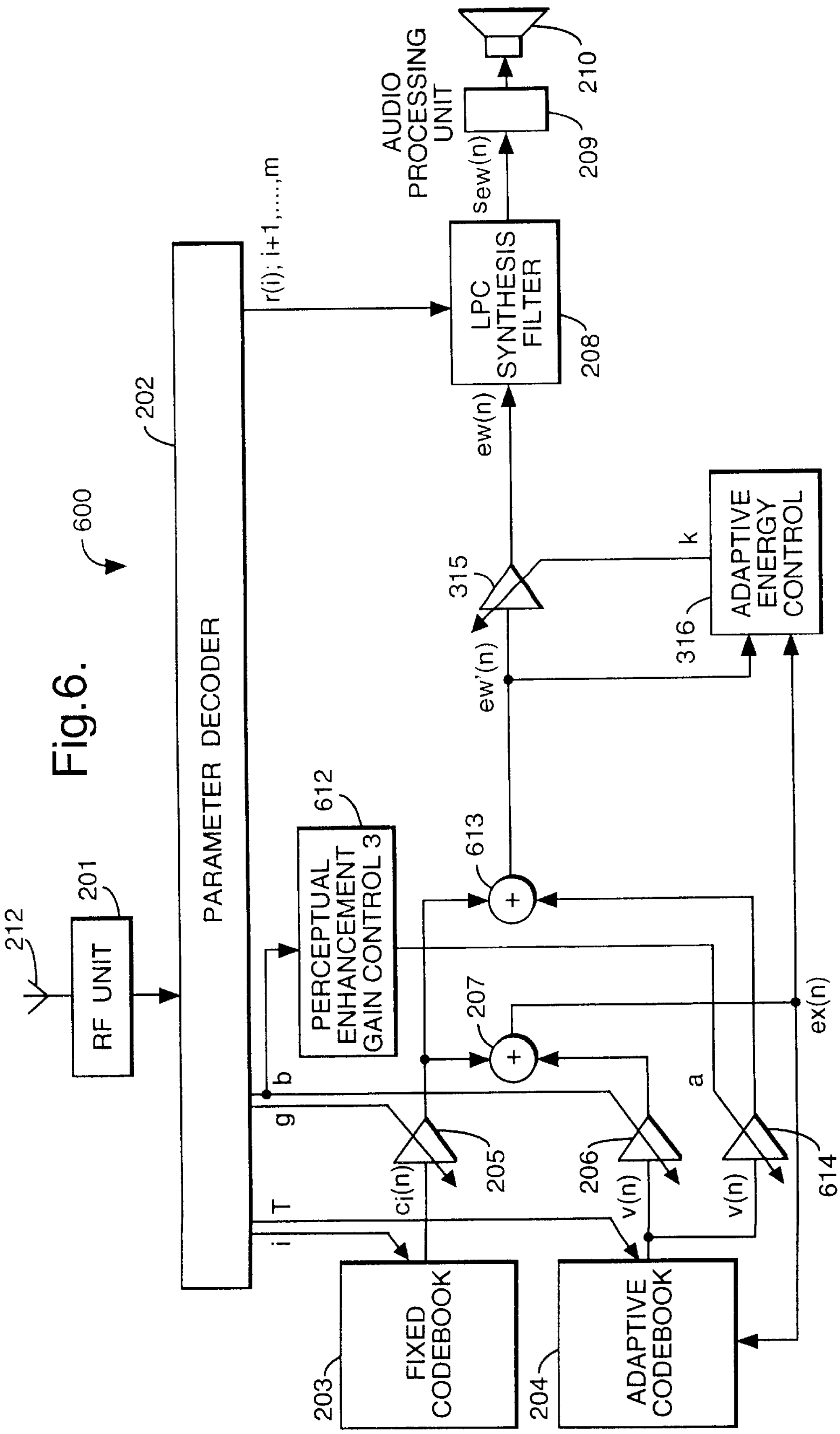
212
RF UNIT
201
PARAMETER DECODER
202
r(i); i=1,...,m
LPC SYNTHESIS FILTER
208
sew(n)
AUDIO PROCESSING UNIT
209
210
ew(n)
315
k
ADAPTIVE ENERGY CONTROL
316
ew'(n)
317
PERCEPTUAL ENHANCEMENT GAIN CONTROL 2
512
p'
313
314
ex(n)
207
b
g
T
i
205
206
ci(n)
v(n)
FIXED CODEBOOK
203
ADAPTIVE CODEBOOK
204
211

Fig.5.



600
212
RF UNIT
201
PARAMETER DECODER
202
PERCEPTUAL ENHANCEMENT GAIN CONTROL 3
612
r(i); i+1,...,m
LPC SYNTHESIS FILTER
208
sew(n)
AUDIO PROCESSING UNIT
209
210
ew(n)
315
k
ADAPTIVE ENERGY CONTROL
316
ew'(n)
613
207
ex(n)
b
g
a
T
i
205
206
614
ci(n)
v(n)
v(n)
FIXED CODEBOOK
203
ADAPTIVE CODEBOOK
204

Fig.6.

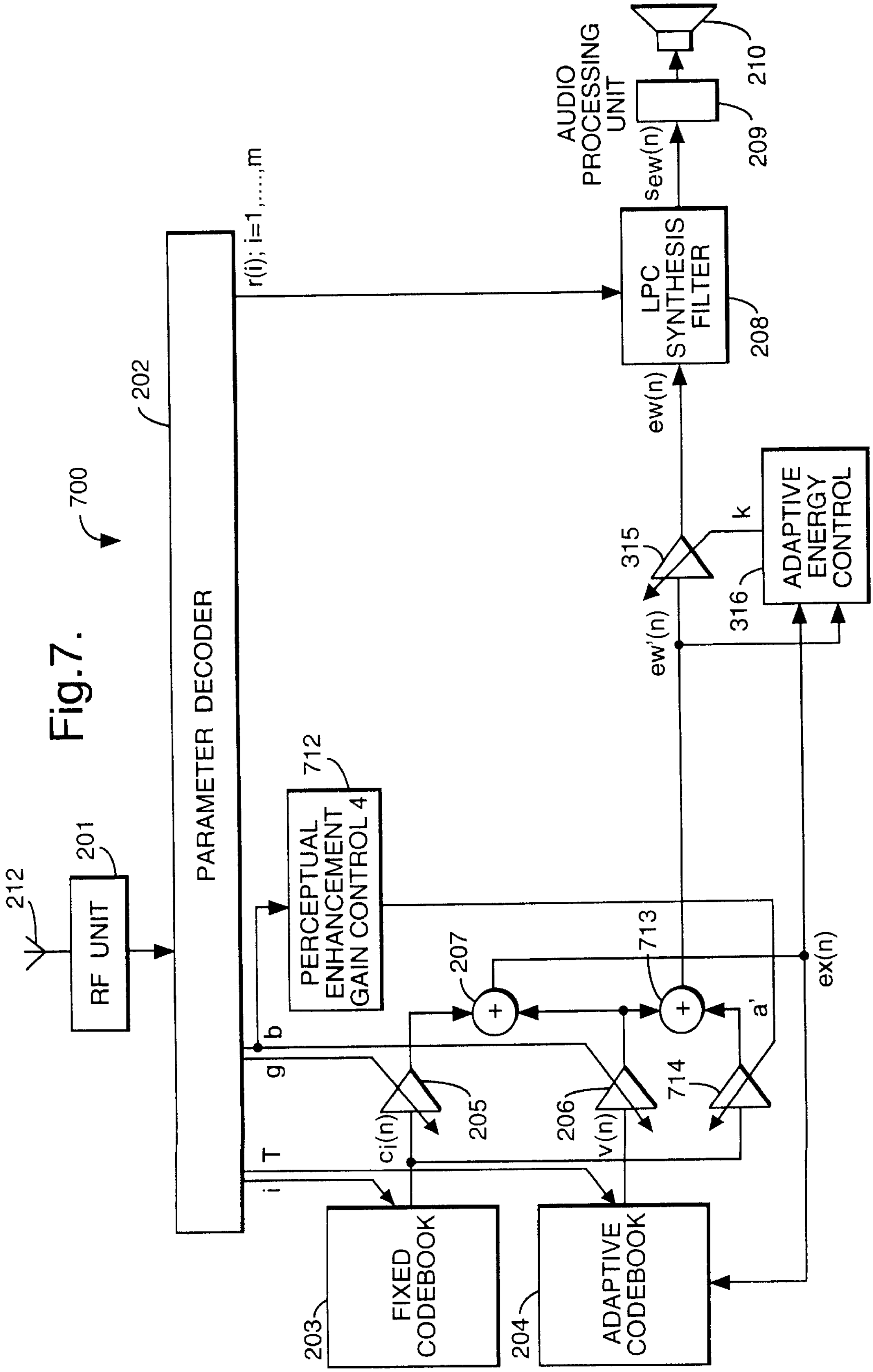

700
212
RF UNIT
201
PARAMETER DECODER
202
r(i); i=1,....,m
LPC SYNTHESIS FILTER
208
sew(n)
AUDIO PROCESSING UNIT
209
210
ew(n)
315
k
ADAPTIVE ENERGY CONTROL
316
ew'(n)
PERCEPTUAL ENHANCEMENT GAIN CONTROL 4
712
207
713
ex(n)
b
g
T
i
a'
205
206
714
ci(n)
v(n)
FIXED CODEBOOK
203
ADAPTIVE CODEBOOK
204

Fig.7.

SPEECH SYNTHESIZER EMPLOYING POST-PROCESSING FOR ENHANCING THE QUALITY OF THE SYNTHESIZED SPEECH

This application is a continuation of copending U.S. patent application Ser. No. 08/662,991, filed Jun. 13, 1996, which in turn claims priority from U.K. Patent Application No.: 9512284, filed on Jun. 15, 1995 (as does this continuation application).

FIELD OF INVENTION

The present invention relates to an audio or speech synthesiser for use with compressed digitally encoded audio or speech signals. In particular, to a post-processor for processing signals derived from an excitation code book and adaptive code book of a LPC type speech decoder.

BACKGROUND TO INVENTION

In digital radio telephone systems the information, i.e. speech, is digitally encoded prior to being transmitted over the air. The encoded speech is then decoded at the receiver. First, an analogue speech signal is digitally encoded using Pulse Code Modulation (PCM) for example. Then speech coding and decoding of the PCM speech (or original speech) is implemented by speech coders and decoders. Due to the increase in use of radio telephone systems the radio spectrum available for such systems is becoming crowded. In order to make the best possible use of the available radio spectrum, radio telephone systems utilise speech coding techniques which require low numbers of bits to encode the speech in order to reduce the bandwidth required for the transmission. Efforts are continually being made to reduce the number of bits required for speech coding to further reduce the bandwidth required for speech transmission.

A known speech coding/decoding method is based on linear predictive coding (LPC) techniques, and utilises analysis-by-synthesis excitation coding. In an encoder utilising such a method, a speech sample is first analysed to derive parameters which represent characteristics such as wave form information (LPC) of the speech sample. These parameters are used as inputs to short-term synthesis filter. The short-term synthesis filter is excited by signals which are derived from a code book of signals. The excitation signals may be random, e.g. a stochastic code book, or may be adaptive or specifically optimised for use in speech coding. Typically, the code book comprises two parts, a fixed code book and the adaptive code book. The excitation outputs of respective code books are combined and the total excitation input to the short term synthesis filter. Each total excitation signal. is filtered and the result compared with the original speech sample (PCM coded) to derive an "error" or difference between the synthesised speech sample and the original speech sample. The total excitation which results in the lowest error is selected as the excitation for representing the speech sample. The code book indices, or addresses, of the location of respective partial optimal excitation signals in the fixed and adaptive code book are transmitted to a receiver, together with the LPC parameters or coefficients. A composite code book identical to that at the transmitter is also located at the receiver, and the transmitted code book indices and parameters are used to generate the appropriate total excitation signal from the receiver's code book. This total excitation signal is then fed to a short-term synthesis filter identical to that in the transmitter, and having the transmitted LPC coefficients as respective inputs. The output from the short-term synthesis filter is a synthesised speech

frame which is the same as that generated in the transmitter by the analysis-by-synthesis method.

Due to the nature of digital coding, although the synthesised speech is objectively accurate it sounds artificial. Also, degradations, distortions and artifacts are introduced into the synthesised speech due to quantisation effects and other anomalies due to the electronic processing. Such artifacts particularly occur in low bit-rate coding since there is insufficient information to reproduce the original speech signal exactly. Hence there have been attempts to improve the perceptual quality of synthesised speech. This has been attempted by the use of post-filters which operate on the synthesised speech sample to enhance its perceived quality. Known post-filters are located at the output of the decoder and process the synthesised speech signal to emphasise or attenuate what are generally considered to be the most important frequency regions in speech. The importance of respective regions of speech frequencies has been analysed primarily using subjective tests on the quality of the resulting speech signal to the human ear. Speech can be split into two basic parts, the spectral envelope (formant structure) or the spectral harmonic structure (line structure), and typically post-filtering emphasises one or other, or both of these parts of a speech signal. The filter coefficients of the post-filter are adapted depending on the characteristics of the speech signal to match the speech sounds. A filter emphasising or attenuating the harmonic structure is typically referred to as a long-term, or pitch or long delay post filter, and a filter emphasising the spectral envelope structure is typically referred to as a short delay post filter or short-term post filter. A further known filtering technique for improving the perceptual quality of synthesised speech is disclosed in International Patent Application WO 91/06091. A pitch prefilter is disclosed in WO 91/06091 comprising a pitch enhancement filter, normally disposed at a position after a speech synthesis or LPC filter, moved to a position before the speech synthesis or LPC filter where it filters pitch information contained in the excitation signals input to the speech synthesis or LPC filter. However, there is still a desire to produce synthesised speech which has even better perceptual quality.

SUMMARY OF INVENTION

According to a first aspect of the present invention there is provided a synthesiser for speech synthesis, comprising a post-processing means for operating on a first signal including speech periodicity information and derived from an excitation source, wherein the post-processing means is adapted to modify the speech periodicity information content of the first signal in accordance with a second signal derivable from the excitation source.

According to a second aspect of the present invention there is provided a method for enhancing synthesised speech, comprising

deriving a first signal including speech periodicity information from an excitation source,

deriving a second signal from the excitation source, and

modifying the speech periodicity information content of the first signal in accordance with the second signal.

An advantage of the present invention is that the first signal is modified by a second signal originating from the same source as the first signal, and thus no additional sources of distortion or artifacts such as extra filters are introduced. Only the signals generated in the excitation source are utilised. The relative contributions of the signals inherent to the excitation generator in a speech synthesiser are being modified, with no artificial added signals, to re-scale the synthesiser signals.

Good speech enhancement may be obtained if post-processing of the excitation is based on modifying the relative contributions of the excitation components derived within the excitation generator of the speech synthesiser itself.

Processing the excitation by filtering the total excitation ex(n) without considering or modifying the relative contributions of the signals inherent to the excitation generator, i.e. v(n) and $c_i(n)$ typically does not give the best possible enhancement. Modifying the first signal in accordance with the second signal from the same excitation source increases waveform continuity within the excitation and in the resulting synthesised speech signal, thereby improving its perceptual quality.

In a preferred embodiment the excitation source comprises a fixed code book and an adaptive code book, the first signal being derivable from a combination of first and second partial excitation signals respectively selectable from the fixed and adaptive code books, which is a particularly convenient excitation source for a speech synthesiser.

Preferably, there is a gain element for scaling the second signal in accordance with a scaling factor (p) derivable from pitch information associated with the first signal from the excitation source, which has the advantage that the first signal speech periodicity information content is modified which has greater effect on perceived speech quality than other modifications.

Suitably, the scaling factor (p) is derivable from an adaptive code book scaling factor (b), and the scaling factor (p) is derivable in accordance with the following equation,

$$
\text{if}\quad
\begin{array}{ll}
b < TH_{low} & \text{then } p = 0.0 \\
TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\
TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\
\vdots & \vdots \\
TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\
b > TH_{upper} & \text{then } p = a_{enhN} f_N(b)
\end{array}
$$

where TH represents threshold values, b is the adaptive code book gain factor, p is the post-processor means scale factor, a_{snh} is a linear scaler and f(b) is a function of gain b

In a specific embodiment the scaling factor (p) is derivable in accordance with

$$
\mathit{if}\quad
\begin{array}{ll}
b < TH_{low} & \text{then } p = 0.0 \\
TH_{low} \le b \le TH_{upper} & \text{then } p = a_{enh} b^2 \\
b > TH_{upper} & \text{then } p = a_{enh} b
\end{array}
$$

where a_{enh} is a constant that controls the strength of the enhancement operation, b is adaptive code book gain, TH are threshold values and p is the post-processor scale factor which utilises the insight that speech enhancement is most effective for voiced speech where b typically has a high value, whereas for unvoiced sounds where b has a low value a not so strong enhancement is required.

The second signal may originate from the adaptive code book, and may also be substantially the same as the second partial excitation signal. Alternatively, the second signal may originate from the fixed code book, and may also be substantially the same as the first partial excitation signal.

For the second signal originating from the fixed code book, the gain control means is adapted to scale the second signal in accordance with a second scaling factor (p')

where,

$$p' = -\frac{gp}{(p+b)}$$

and g is a fixed code book scaling factor, b is an adaptive code book scaling factor and p is the first scaling factor.

The first signal may be a first excitation signal suitable for inputting to a speech synthesis filter, and the second signal may be a second excitation signal suitable for inputting to a speech synthesis filter. The second excitation signal may be substantially the same as the second partial excitation signal.

Optionally, the first signal may be a first synthesised speech signal output from a first speech synthesis filter and derivable from the first excitation signal, and the second signal may be the output from a second speech synthesis filter and derivable from the second excitatiori signal. An advantage of this is that speech enhancement is carried out on the actual synthesised speech and thus there are less electronic components to introduce distortion to the signal before it is rendered audible.

Advantageously, there is provided an adaptive energy control means adapted to scale a modified first signal in accordance with the following relationship,

$$k = \sqrt{\frac{\sum_{n=0}^{N-1} ex^2(n)}{\sum_{n=0}^{N-1} ew'^2(n)}}$$

where N is a suitably chosen adaption period, ex(n) is the first signal, ew'(n) is the modified first signal and k is an energy scale factor. which normalises the resulting enhanced signal to the power input to the speech synthesiser.

In a third aspect according to the invention there is provided, a radio device, comprising

a radio frequency means for receiving a radio signal and recovering coded information included in the radio signal, and

an excitation source coupled to the radio frequency means for generating a first signal including speech periodicity information in accordance with the coded information, wherein the radio device further comprises a post-processing means operably coupled to the excitation source to receive the first signal and adapted to modify the speech periodicity information content of the first signal in accordance with a second signal derived from the excitation source and a speech synthesis filter coupled to receive the modified first signal from the post-processing means and for generating synthesised speech in response thereto.

In a fourth aspect of the invention there is provided a synthesiser for speech synthesis, comprising first and second excitation sources for respectively generating first and second excitation signals, and modifying means for modifying the first excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal.

In a fifth aspect of the invention there is provided a synthesiser for speech synthesis, comprising first and second excitation sources for respectively generating first and second excitation signals, and modifying means for modifying the second excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal.

The fourth and fifth aspects of the invention advantageously integrate scaling of excitation signals within the excitation generator itself.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** shows a schematic diagram of a known Code Excitation Linear Prediction (CELP) encoder;

FIG. **2** shows a schematic diagram of a known CELP decoder;

FIG. **3** shows a schematic diagram of a CELP decoder in accordance with a first embodiment of the invention;

FIG. **4** shows a second embodiment in accordance with the invention;

FIG. **5** shows a third embodiment in accordance with the invention;

FIG. **6** shows a fourth embodiment in accordance with the invention; and

FIG. **7** shows a fifth embodiment in accordance with the invention.

DETAILED DESCRIPTION OF EMBODIMENTS OF THE INVENTION

Embodiments in accordance with the invention will now be described, by way of example only, and with reference to the accompanying drawings.

A known CELP encoder **100** is shown in FIG. **1**. Original speech signals are input to the encoder at **102** and Long Term Prediction (LTP) coefficients T,b are determined using adaptive code book **104**. The LTP prediction coefficients are determined for segments of speech typically comprising 40 samples and are 5 ms in length. The LTP coefficients relate to periodic characteristics of the original speech. This includes any periodicity in the original speech and not just to periodicity which corresponds to the pitch of the original speech due to vibrations in the vocal cords of a person uttering the original speech.

Long Term Prediction is performed using adaptive code book **104** and gain element **114**, which comprise a part of excitation signal (ex(n)) generator **126** shown dotted in FIG. **1**. Previous excitation signals ex(n) are stored in the adaptive code book **104** by virtue of feedback loop **122**. During the LTP process the adaptive code book is searched by varying an address T, known as a delay or lag, pointing to previous excitation signals ex(n). These signals are sequentially output and amplified at gain element **114** with a scaling factor b to form signals v(n) prior to being added at **118** to an excitation signal $c_i(n)$ derived from the fixed code book **112** and scaled by a factor g at gain element **116**. Linear Prediction Coefficients (LPC) for the speech sample are calculated at **106**. The LPC coefficients are then quantised at **108**. The quantised LPC coefficients are then available for transmission over the air and to be input to short term filter **110**. The LPC coefficients (r(i), i=1 . . . , m where m is prediction order) are calculated for segments of speech comprising 160 samples over 20 ms. All further processing is typically performed in segments of 40 samples, that is to say an excitation frame length of 5 ms. The LPC coefficients relate to the spectral envelope of the original speech signal.

Excitation generator **126** effectively comprises a composite code book **104, 112** comprising sets of codes for exciting short term synthesis filter **110**. The codes comprise sequences of voltage amplitudes, each corresponding to a speech sample in the speech frame.

Each total excitation signal ex(n) is input to short term or LPC synthesis filter **110** to form a synthesised speech sample s(n). The synthesised speech sample s(n) is input to a negative input of adder **120**, having an original speech sample as a positive input. The adder **120** outputs the difference between the original speech sample and the synthesised speech sample, this difference being known as an objective error. The objective error is input to a best excitation selection element **124**, which selects the total excitation ex(n) resulting in a synthesised speech frame s(n) having the least objective error. During the selection the objective error is typically further spectrally weighted to emphasise those spectral regions of the speech signal important for human perception. The respective adaptive and fixed code book parameters (gain b and delay T, and gain g and index i) giving the best excitation signal ex(n) are then transmitted, together with the LPC filter coefficients r(i), to a receiver to be used in synthesising the speech frame to reconstruct the original speech signal.

A decoder suitable for decoding speech parameters generated by an encoder as described with reference to FIG. **1** is shown in FIG. **2**. Radio frequency unit **201** receives a coded speech signal via an antenna **212**. The received radio frequency signal is down converted to a baseband frequency and demodulated in the RF unit **201** to recover speech information. Generally, coded speech is further encoded prior to being transmitted to comprise channel coding and error correction coding. This channel coding and error correction coding has to be decoded at the receiver before the speech coding can be accessed or recovered. Speech coding parameters are recovered by parameter decoder **202**.

The speech coding parameters in LPC speech coding are the set of LPC synthesis filter coefficients r(i); i=1 . . . ,m, (where m is the order of the prediction), fixed code book index i and gain g. The adaptive code book speech coding parameters delay T and gain b are also recovered.

The speech decoder **200** utilises the above mentioned speech coding parameters to create from the excitation generator **211** an excitation signal ex(n) for inputting to the LPC synthesis filter **208** which provides a synthesised speech frame signal s(n) at its output as a response to the excitation signal ex(n). The synthesised speech frame signal s(n) is further processed in audio processing unit **209** and rendered audible through an appropriate audio transducer **210**.

In typical linear predictive speech decoders, the excitation signal ex(n) for the LPC synthesis filter **208** is formed in excitation generator **211** comprising a fixed code book **203** generating excitation sequence $c_i(n)$ and adaptive code book **204**. The location of the code book excitation sequence ex(n) in the respective code books **203**, **204** is indicated by the speech coding parameter i and delay T. The fixed code book excitation sequence $c_i(n)$ partially used to form the excitation signal ex(n) is taken from the fixed excitation code book **203** from a location indicated by index i and is then suitably scaled by the transmitted gain factor g in the scaling unit **205**. Similarly, the adaptive code book excitation sequence v(n) also partially used to form excitation signal ex(n) is taken from the adaptive code book **204** from a location indicated by delay T using selection logic inherent to the adaptive code book and is then suitably scaled by the transmitted gain factor b in scaling unit **206**.

The adaptive code book **204** operates on the fixed code book excitation sequence $c_i(n)$ by adding a second partial excitation component v(n) to the code book excitation sequence g $c_i(n)$. The second component is derived from past excitation signals in a manner already described with reference to FIG. **1**, and is selected from the adaptive code book **204** using selection logic suitably included in the adaptive code book. The component v(n) is suitably scaled in the scaling unit **206** by the transmitted adaptive code book gain b and then added to g $c_i(n)$ in the adder **207** to form the total excitation signal ex(n), where

$$\mathrm{ex}(n)=g\ c_i(n)+b\ v(n). \tag{1}$$

The adaptive code book **204** is then updated by using the total excitation signal ex(n).

The location of the second partial excitation component v(n) in the adaptive code book **204** is indicated by the speech coding parameter T. The adaptive excitation component is selected from the adaptive code book using speech coding parameter T and selection logic included in the adaptive code book.

An LPC speech synthesis decoder **300** in accordance with the invention is shown in FIG. **3**. The operation of speech synthesis according to FIG. **3** is the same as for FIG. **2** except that the total excitation signal ex(n) is, prior to being used as the excitation for the LPC synthesis filter **208**, processed in excitation post-processing unit **317**. The operation of circuit elements **201** to **212** in FIG. **3** are similar to those in FIG. **2** with the same numerals.

In accordance with an aspect of the invention, a post-processing unit **317** for the total excitation ex(n) is used in the speech decoder **300**. The post-processing unit **317** comprises an adder **313** for adding a third component to the total excitation ex(n). A gain unit **315** then appropriately scales the resulting signal ew'(n) to form signal ew(n) which is then used to excite the LPC synthesis filter **208** to produce synthesised speech signal $s_{ew}(n)$. The speech synthesised according to the invention has improved perceptual quality compared to the speech signal s(n) synthesised by the prior art speech synthesis decoder shown in FIG. **2**.

The post-processing unit **317** has the total excitation ex(n) input to it, and outputs a perceptually enhanced total excitation ew(n). The post-processing unit **317** also has. the adaptive code book gain b, and an unscaled partial excitation component v(n) taken from the adaptive code book **204** at a location indicated by the speech coding parameters as further inputs. Partial excitation component v(n) is suitably the same component which is employed inside the excitation generator **211** to form the second excitation component bv(n) which is added to the scaled code book excitation $gc_i(n)$ to form the total excitation ex(n). By using an excitation sequence which is derived from the adaptive code book **204**, no further sources of artifacts are added to the speech processing electronics, as is the case with the known post or pre-filter techniques which use extra filters. The excitation post-processing unit **317** also comprises scaling unit **314** which scales the partial excitation component v(n) by a scale factor p, and the scaled component pv(n) is added by adder **313** to the total excitation component ex(n). The output of adder **313** is an intermediate total excitation signal ew'(n). it is of the form,

$$ew'(n)=gc_i(n)+bv(n)+pv(n)=gc_i(n)+(b+p)\ v(n). \tag{2}$$

The scaling factor p for scaling unit **314** is determined in the perceptual enhancement gain control unit **312** using the adaptive code book gain b. The scaling factor pre-scales the contribution of the two excitation components from the fixed and adaptive code book, $c_i(n)$ and v(n), respectively. The scaling factor p is adjusted so that during synthesised speech frame samples that have high adaptive code book gain value b the scale factor p is increased, and during speech that has low adaptive code book gain value b the scaling factor p is reduced. Furthermore, when b is less than a threshold value

($b<TH_{low}$) the scaling factor p is set to zero. The perceptual enhancement gain control unit **312** operates in accordance with equation (3) given below,

$$\text{if}\ \begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b \le TH_{upper} & \text{then } p = a_{enh} b^2 \\ b > TH_{upper} & \text{then } p = a_{enh} b \end{array} \tag{3}$$

where a_{enh} is a constant that controls the strength of the enhancement operation. The applicant has found that a good value for a_{enh} is 0.25, and good values for TH_{low} and TH_{upper} are 0.5 and 1.0, respectively.

Equation 3 can be of a more general form, and a general formulation of the enhancement function is shown below in equation (4). In the general case, there could be more than 2 thresholds for the enhancement gain b. Also, the gain could be defined as a more general function of b.

$$\text{if}\ \begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array} \tag{4}$$

In the preferred embodiment previously described N=2, TH_{low}=0.5, TH_2=1.0, TH3= , a_{enh1}=0.25, and a_{enh2}=0.25, $f_1(b)=b^2$, and $f^2(b)=b$.

The threshold values (TH), enhancement values (a_{enh}) and the gain functions (f(b)) are arrived at empirically. Since the only realistic measure of perceptual speech quality can be obtained by human beings listening to the speech and giving their subjective opinions on the speech quality, the values used in equations (3) and (4) are determined experimentally. Various values for the enhancement thresholds and gain functions are tried, and those resulting in the best sounding speech are selected. The applicant has utilised the insight that the enhancement to the speech quality using this method is particularly effective for voiced speech where b typically has a high value, whereas for less voiced sounds which have a lower value of b not so strong an enhancement is required. Thus, gain value p is controlled such that for voiced sounds, where the distortions are most audible, the effect is strong and for unvoiced sounds the effect is weaker or not used at all. Thus, as a general rule, the gain functions (f_n) should be chosen so that there is a greater effect for higher values of b, than for lower values of b. This increases the difference between the pitch components of the speech and the other components.

In the preferred embodiment, operating in accordance with equation (3), the functions operating on gain value b are a squared dependency for mid-range values of b and a linear dependency for high-range values of b. it is the applicant's present understanding that this gives good speech quality since for high values of b, i.e. highly voiced speech, there is greater effect and for lower values of b there is less effect. This is because b typically lies in the range $-1<b<1$ and therefore $b^2<b$.

To ensure unity power gain between the input signal ex(n), and the output signal ew(n) of the excitation post-processing unit **317**, a scale factor is computed and is used to scale the intermediate excitation signal ew'(n) in the scaling unit **315** to form the post-processed excitation signal ew(n). The scale factor k is given as

$$k = \sqrt{\frac{\sum_{n=0}^{N-1} ex^2(n)}{\sum_{n=0}^{N-1} ew'^2(n)}} \tag{5}$$

where N is a suitably chosen adaption period. Typically, N is set equal to the excitation frame length of the LPC speech codec.

In the adaptive code book of the encoder, for values of T which are less than the frame length or excitation length a part of the excitation sequence is unknown. For these unknown portions a replacement sequence is locally generated within the adaptive code book by using suitable selection logic. Several adaptive code book techniques to generate this replacement sequence are known from the state of the art. Typically, a copy of a portion of the known excitation is copied to where the unknown portion is located thereby creating a complete excitation sequence. The copied portion may be adapted in some manner to improve the quality of the resulting speech signal. When doing such copying, the delay value Tis not used since it would point to the unknown portion. Instead, a particular selection logic resulting in a modified value for T is used (for example, using T multiplied by an integer factor so that it always points to the known signal portion). So that the decoder is synchronised with the encoder, similar modifications are employed in the adaptive code book of the decoder. By using such a selection logic to generate a replacement sequence within the adaptive code book, the adaptive code book is able to adapt for high pitch voices such as female and child voices resulting in efficient excitation generation and improved speech quality for these voices.

For obtaining good perceptual enhancement, all modifications inherent to the adaptive code book e.g. for values of T less than the frame length are taken into account in the enhancement post-processing. This is obtained in accordance with the invention by the use of the partial excitation sequence from the adaptive code book v(n) and the re-scaling of the excitation components, inherent to the excitation generator of the speech synthesiser.

In summary, the method enhances the perceptual quality of the synthesised speech and reduces audible artifacts by adaptively scaling the contribution of the partial excitation components taken from the code book **203** and from the adaptive code book **204**, in accordance with equations (2), (3), (4) and (5).

FIG. **4** shows a second embodiment in accordance with the invention, wherein the excitation post-processing unit **417** is located after the LPC synthesis filter **208** as illustrated. In this embodiment an additional LPC synthesis filter **408** is required for the third excitation component derived from the adaptive code book **204**. In FIG. **4**, elements which have the same function as in FIGS. **2** and **3**, also have the same reference numerals.

In the second embodiment shown in FIG. **4**, the LPC synthesised speech is perceptually enhanced by post-processor **417**. The total excitation signal ex(n) derived from the code book **203** and adaptive code book **204** is input to LPC synthesis filter **208** and processed in a conventional manner in accordance with the LPC coefficients r(i). The additional or third partial excitation component v(n) derived from the adaptive code book **204** in the manner described in relation to FIG. **3** is input unscaled to a second LPC synthesis filter **408** and processed in accordance with the LPC coefficients r(i). The outputs s(n) and $s_v(n)$ of respective LPC filters **208**, **408** are input to post-processor **417** and added together in adder **413**. Prior to being input to adder **413**, signal $s_v(n)$ is scaled by scale factor p. As described with reference to FIG. **3**, the values for processing scale factor or gain p can be arrived at empirically. Additionally, the third partial excitation component may be derived from the fixed code book **203** and the scaled speech signal $p's_v(n)$ subtracted from speech signal s(n).

The resulting perceptually enhanced output $s_w(n)$ is then input to the audio processing unit **209**.

Optionally, a further modification of the enhancement system can be formed by moving the scaling unit **414** of FIG. **4** to be in front of the LPC synthesis filter **408**. Locating the post-processor **417** after the LPC or short term synthesis filters **208**, **408** can give better control of the emphasis of the speech signal since it is carried out directly on the speech signal, not on the excitation signal. Thus, less distortions are likely to occur.

Optionally, enhancement can be achieved by modifying the embodiments described with reference to FIGS. **3** and **4** respectively, such that the additional (third) excitation component is derived from the fixed code book **203** instead of the adaptive code book **204**. Then, a negative scaling factor should be used instead of the original positive gain factor p, to decrease the gain for excitation sequence $c_i(n)$ from the fixed code book. This results in a similar modification of the relative contributions of the partial excitation signals $c_i(n)$ and v(n), to speech synthesis as achieved with the embodiments of FIGS. **3** and **4**.

FIG. **5** shows an embodiment in accordance with the invention in which the same result as obtained by using scaling factor p and the additional excitation component from the adaptive code book may be achieved. In this embodiment, the fixed code book excitation sequence $c_i(n)$ is input to scaling unit **314** which operates in accordance with scale factor p' output from perceptual enhancement gain control **2 512**. The scaled fixed code book excitation, p' $c_i(n)$, output from scaling unit **314** is input to adder **313** where it is added to total excitation sequence ex(n) comprising components $c_i(n)$ and v(n) from the fixed code book **203** and adaptive code book **204** respectively.

When increasing the gain for the excitation sequence signal v(n) from the adaptive code book **204** the total excitation (before adaptive energy control **316**) is given by equation (2), viz.

$$ew'(n)=g\,c_i(n)+(b+p)\,v(n) \tag{2}$$

When decreasing the gain for an excitation sequence $c_i(n)$ from the fixed code book **203**, the total excitation (before adaptive energy control **316**) is given as

$$ew'(n)=(g+p')\,c_i(n)+bv(n) \tag{6},$$

where p' is the scaling factor derived by perceptual enhancement gain control **2 512** shown in FIG. **5**. Taking equation (2) and reformulating it into a form similar to equation (6) gives:

$$\begin{aligned} ew'(n) &= g\,c_i(n)+(b+p)\,v(n) \\ &= \frac{p+b}{b}\left[\left(\frac{gb}{p+b}\right)c_i(n)+bv(n)\right] \\ &= \frac{p+b}{b}\left[\left(g-\frac{gp}{p+b}\right)c_i(n)+bv(n)\right] \end{aligned}$$

Thus, selecting

$$p' = -\frac{gp}{(p+b)}$$

In the embodiment of FIG. **5** a similar enhancement as obtained with the embodiment of FIG. **3** will be achieved. When the intermediate total excitation signal ew'(n) is scaled by adaptive energy control **316** to the same energy content as ex(n), then both embodiments, FIG. **3** and FIG. **5**, result in the same total excitation signal ew(n).

Perceptual enhancement gain control **2 512** can therefore utillse the same processing as employed in relation to the embodiments of FIGS. **3** and **4** to generate "p", and then utilise equation (8) to get p'.

The intermediate total excitation signal ew'(n) output from adder **313**. is scaled in scaling unit **315** under control of adaptive energy control **316** in a similar manner as described above in relation to the first and second embodiments.

Referring now to FIG. **4**, LPC synthesised speech may be perceptually enhanced by post-processor **417** by synthesised speech derived from additional excitation signals from the fixed code book.

The dotted line **420** in FIG. **4** shows an embodiment wherein the fixed code book excitation signals $c_i(n)$ are coupled to LPC synthesis filter **408**. The output of the LPC synthesis filter **408** ($sc_i(n)$) is then scaled in unit **414** in accordance with scaling factor p' derived from perceptual enhancement gain control **512**, and added to the synthesised signal s(n) in adder **413** to produce intermediate synthesis signal $s'_w(n)$. After normalisation in scaling unit **415** the resulting synthesis signal $s_w(n)$ is forwarded to the audio processing unit **209**.

The foregoing embodiments comprise adding a component derived from the adaptive code book **204** or fixed code book **203** to an excitation ex(n) or synthesised s(n), to form an intermediate excitation ew'(n) or synthesised signal s'_w (n).

Optionally, post-processing may be dispensed with and the adaptive code book v(n) or fixed code book c(n) excitation signals may be scaled and directly combined together. Thereby obviating the addition of components to unscaled combined fixed and adaptive code book signals.

FIG. **6** shows an embodiment in accordance with an aspect of the invention having the adaptive code book excitation signals v(n) scaled and then combined with the fixed code book excitation signals $c_i(n)$ to directly form an intermediate signal ew(n). Perceptual enhancement gain control **612** outputs parameter "a" to control scaling unit **614**. Scaling unit **614** operates on adaptive code book excitation signal v(n) to scale-up or amplify excitation signal v(n) over the gain factor b used to get the normal excitation. Normal excitation ex(n) is also formed and coupled to the adaptive code book **204** and adaptive energy control **316**. Adder **613** combines up-scaled excitation signal av(n) and fixed code book excitation $c_i(n)$ to form an intermediate signal;

$$ew'(n)=g\ c_i(n)+av(n) \tag{9}$$

If a=b+p, then the same processing as given by equation (2) may be achieved.

FIG. **7** shows an embodiment operable in a manner similar to that shown in FIG. **6**, but down-scaling or attenuating the fixed code book excitation signal $c_i(n)$. For this embodiment the intermediate excitation sign ew'(n) is given by:

$$ew'(n)=(g+p')\ c_i(n)+bv(n)=a'c_i(n)+bv(n) \tag{10}$$

where,

$$a' = g - \frac{gp}{p+b} = \frac{gb}{p+b}. \tag{11}$$

Perceptual enhancement gain control **712** outputs a control signal a' in accordance with equation (11), to obtain a similar result as obtained with equation (6) in accordance with equation (8). The down-scaled fixed code book excitation signal $a'c_i(n)$ is combined with adaptive code book excitation signal v(n) in adder **713** to form intermediate excitation signal ew'(n). The remaining processing is carried out as described before, to normalise the excitation signal and formed synthesised signal $s_{ew}(n)$.

The embodiments described with reference to FIGS. **6** and **7** perform scaling of the excitation signals within the excitation generator, and directly from the code books.

The determination of scaling factor "p" for the embodiments described with reference to FIGS. **5**, **6** and **7** may be made in accordance with equations (3) or (4) described above.

Various methods of control of the enhancement level (a_{enh}) may be employed. In addition to the adaptive code book gain b, the amount of enhancement could be a function of the lag or delay value T for the adaptive code book **204**. For example, the post processing could be turned on (or emphasised) when operating in a high pitch range or when the adaptive code book parameter T is shorter than the excitation block length (virtual lag range). As a result, female and child voices, forwhich the invention is most beneficial, would be highly post processed.

The post processing control could also be based on voiced/unvoiced speech decisions. For example, the enhancement could be stronger for voiced speech, and it could be totally turned off when the speech is classified as unvoiced. This can be derived from the adaptive code book gain value b which is itself a simple measure of voiced/unvoiced speech, that is to say the higher b, the more voiced speech present in the original speech signal.

Embodiments in accordance with the present invention may be modified, such that the third partial excitation sequence is not the same partial excitation sequence derived from the adaptive code book or fixed code book in accordance with conventional speech synthesis, but is selectable via selection logic typically included in respective code books to choose another third partial excitation sequence. The third partial excitation sequence may be chosen to be the immediately previously used excitation sequence or to be always a same excitation sequence stored in the fixed code book. This would act to reduce the difference between speech frames and thereby enhance the continuity of the speech. Optionally, b and/or T can be recalculated in the decoder from the synthesised speech and used to derive a third partial excitation sequence. Further, a fixed gain p and/or fixed excitation sequence can be added or subtracted as appropriate to the total excitation sequence ex(n) or speech signal s(n) depending on the location of the post-processor.

In view of the foregoing description it will be evident to a person skilled in the art that various modifications may be made within the scope of the invention. For example, variable-frame-rate coding, fast code book searching, reversal of the order of pitch prediction and LPC prediction may be utilised in the codec. Additionally, post-processing in accordance with the present invention could also be included in the encoder, not just the decoder. Furthermore, aspects of respective embodiments described with reference to the drawings may be combined to provide further embodiments in accordance with the invention.

The scope of the present disclosure includes any novel feature or combination of features disclosed therein either explicitly or implicitly or any generalisatlon thereof irrespective of whether or not it relates to the claimed invention or mitigates any or all of the problems addressed by the present invention. The applicant hereby gives notice that new claims may be formulated to such features during prosecution of this application or of any such further application derived therefrom.

What we claim is:

1. A Linear Predictive Coding (LPC) synthesiser for speech synthesis, comprising:

an excitation source; and

a LPC decoder comprising post-processing means coupled to an output of said excitation source for operating on a first signal including speech periodicity information derived from said excitation source, wherein the post-processing means modifies the speech periodicity information content of the first signal in accordance with a second signal derivable from said excitation source in order to produce an enhanced synthesised speech signal.

2. A synthesiser according to claim **1**, wherein the post-processing means comprises gain control means for scaling the second signal in accordance with a first scaling factor (p) derivable from pitch information associated with the first signal.

3. A synthesiser according to claim **2**, wherein the excitation source comprises a fixed code book and an adaptive code book, the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books.

4. A synthesiser according to claim **3**, wherein the first scaling factor (p) is derivable from an adaptive code book gain factor (b).

5. A synthesiser according to claim **4**, wherein the first scaling factor (p) is derivable in accordance with the following relationship,

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, b is the adaptive code book gain factor, p is the first post-processing means scale factor, a_{enh} is a linear scaler and f(b) is a function of the adaptive code book gain factor b.

6. A synthesiser according to claim **4**, wherein the scaling factor (p) is derivable in accordance with

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b \le TH_{upper} & \text{then } p = a_{enh} b^2 \\ b > TH_{upper} & \text{then } p = a_{enh} b \end{array}$$

where a_{enh} is a constant that controls the strength of the enhancement operation, b is the adaptive code book gain factor, TH are threshold values and p is the first post-processing means scale factor.

7. A synthesiser according to claim **3**, wherein the second signal originates from the adaptive code book.

8. A synthesiser according to claim **7**, wherein the second signal is substantially the same as the second partial excitation signal.

9. A synthesiser according to claim **7**, wherein the first signal is a first excitation signal suitable for inputting to a speech synthesis filter, and the second signal is a second excitation signal suitable for inputting to a speech synthesis filter.

10. A synthesiser according to claim **3**, wherein the second signal originates from the fixed code book.

11. A synthesiser according to claim **10**, wherein the second signal is substantially the same as the first partial excitation signal.

12. A synthesiser according to claim **10**, wherein the gain control means scales the second signal in accordance with a second scaling factor (p') where,

$$p' = -\frac{gp}{(p+b)}$$

and where g is a fixed code book scaling factor, b is an adaptive code book gain factor and p is the first scaling factor.

13. A synthesiser according to claim **12**, wherein the first signal is a first synthesised speech signal output from a first speech synthesis filter, the second signal is the output from a second speech synthesis filter, and the gain control means operates on signals input to the second speech synthesis filter.

14. A synthesiser according to claim **10**, wherein the first signal is a first synthesised speech signal output from a first speech synthesis filter, the second signal is the output from a second speech synthesis filter, and the gain control means operates on signals input to the second speech synthesis filter.

15. A synthesiser according to claim **2**, wherein the excitation source comprises a fixed code book and an adaptive code book, the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books, the second signal being substantially the same as the second partial excitation signal and originating from the adaptive code book, the first signal being modified by combining the second signal with the first signal, and the first scaling factor (p) being derivable from an adaptive code book gain factor (b) in accordance with the following relationship,

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, b is the adaptive code book gain factor, p is the first post-processing means scale factor, a_{enh} is a linear scaler and f(b) is a function of the adaptive code book gain factor b.

16. A synthesiser according to claim **2**, wherein the excitation source comprises a fixed code book and an adaptive code book, the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books, the second signal being substantially the same as the first partial excitation signal and originating from the fixed code book, the first signal being modified by combining the second signal with the first signal, and the first scaling factor (p) being derivable from an adaptive code book gain factor (b) in accordance with the following relationship,

$$\text{if}\quad \begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \leq b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \leq b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \leq b \leq TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, b is the adaptive code book gain factor, p is the first post-processing means scale factor, a_{enh} is a linear scaler and f(b) is a function of the adaptive code book gain factor b.

17. A method for use with Linear Predictive Coding (LPC) for enhancing synthesised speech, comprising steps of:

deriving a first signal including speech periodicity information from an excitation source,

deriving a second signal from the excitation source, and

modifying in a LPC decoder the speech periodicity information content of the first signal in accordance with the second signal in order to produce an enhanced synthesised speech signal.

18. A method according to claim **17**, further comprising scaling the second signal in accordance with a first scaling factor (p) derived from pitch information associated with the first signal.

19. A method according to claim **18**, wherein the excitation source comprises a fixed code book and an adaptive code book, the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books.

20. A method according to claim **19**, wherein the first scaling factor (p) is derivable from a gain factor (b) for the pitch information of the first signal.

21. A method according to claim **20**, wherein the first scaling factor (p) is derivable in accordance with the following relationships,

$$\text{if}\quad \begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \leq b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \leq b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \leq b \leq TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, b is the gain factor for the pitch information of the first signal, p is the first scaling factor, a_{enh} is a linear scaler and f(b) is a function of b.

22. A method according to claim **19**, wherein the second signal originates from the adaptive code book.

23. A method according to claim **22**, wherein the second signal is substantially the same as the second partial excitation signal.

24. A method according to claim **22**, wherein the first signal is a first synthesised speech signal output from a first speech synthesis filter and the second signal is the output of a second speech synthesis filter.

25. A method according to claim **19**, wherein the second signal originates from the fixed code book.

26. A method according to claim **25**, wherein the second signal is substantially the same as the first partial excitation signal.

27. A method according to claim **25**, wherein the second signal is scaled in accordance with a second scaling factor (p') where,

$$p' = -\frac{gp}{(p+b)}$$

g is a fixed code book scaling factor, b is an adaptive code book scaling factor and p is the first scaling factor.

28. A method according to claim **25**, wherein the first signal is a first synthesised speech signal output from a first speech synthesis filter and the second signal is the output of a second speech synthesis filter.

29. A method according to claim **17**, wherein the first signal is a first excitation signal suitable for inputting to a first speech synthesis filter, and the second signal is a second excitation signal suitable for inputting to a second speech synthesis filter.

30. A method for use with Linear Predictive Coding (LPC) for enhancing synthesised speech, comprising steps of:

deriving a first signal including speech periodicity information from an excitation source, comprising a fixed code book and an adaptive code book,

the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books,

deriving a second signal from the excitation source, and

modifying in a LPC decoder the speech periodicity information content of the first signal in accordance with the second signal in order to produce an enhanced synthesised speech signal,

the second signal being substantially the same as the second partial excitation signal and originating from the adaptive code book, the first signal being modified by combining the second signal with the first signal, and a first scaling factor (p) being derivable from an adaptive code book scaling factor (b) in accordance with the following relationship,

$$\text{if}\quad \begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \leq b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \leq b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \leq b \leq TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, a_{enh} is a linear scaler and f(b) is a function of b.

31. A method for use with Linear Predictive Coding (LPC) for enhancing synthesised speech, comprising steps of:

deriving a first signal including speech periodicity information from an excitation source, comprising a fixed code book and an adaptive code book,

the first signal comprising a combination of first and second partial excitation signals respectively originating from the fixed and adaptive code books,

deriving a second signal from the excitation source, and

modifying in a LPC decoder the speech periodicity information content of the first signal in accordance with the second signal in order to produce an enhanced synthesised speech signal,

the second signal being substantially the same as the first partial excitation signal and originating from the fixed code book, the first signal being modified by combining the second signal with the first signal, and a first scaling factor (p) being derivable from an adaptive code book scaling factor (b) in accordance with the following relationship,

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, a_{enh} is a linear scaler and f(b) is a function of b.

32. A Linear Predictive Coding (LPC) synthesiser for speech synthesis, comprising first and second excitation sources for respectively generating first and second excitation signals, and a LPC decoder comprising modifying means for modifying the first excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal in order to produce an enhanced synthesised speech signal.

33. A synthesiser according to claim **32**, wherein the modifying means scales the first excitation signal in accordance with a scaling factor (a) derivable from pitch information associated with the first signal.

34. A synthesiser according to claim **33**, wherein the first excitation source is an adaptive code book and the second excitation source is a fixed code book.

35. A synthesiser according to claim **34**, wherein the scaling factor (a) is of the form a=b+p, where b is an adaptive code book gain and p is a perceptual enhancement gain factor derivable in accordance with the following relationships;

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, a_{enh} is a linear scaler and f(b) is a function of gain b.

36. A synthesiser according to claim **35**, wherein the first and second excitation signals are combined after modification.

37. A synthesiser according to claim **34**, wherein the scaling factor (a) is of the form a=b+p, where b is an adaptive code book gain and p is a perceptual enhancement gain factor, and wherein the perceptual enhancement gain factor p is derivable in accordance with;

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b \le TH_{upper} & \text{then } p = a_{enh} b^2 \\ b > TH_{upper} & \text{then } p = a_{enh} b \end{array}$$

where a_{enh} is a constant that controls the strength of the enhancement operation and TH are threshold values.

38. A Linear Predictive Coding (LPC) synthesiser for speech synthesis, comprising first and second excitation sources for respectively generating first and second excitation signals, and a LPC decoder comprising modifying means for modifying the second excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal in order to produce an enhanced synthesised speech signal.

39. A synthesiser according to claim **38**, wherein the modifying means scales the second excitation signal in accordance with a scaling factor (a') derivable from pitch information associated with the first signal.

40. A synthesiser according to claim **39**, wherein the first excitation source is an adaptive code book and the second excitation source is a fixed code book.

41. A synthesiser according to claim **40**, wherein the scaling factor (a') satisfies the following relationship;

$$a' = -\frac{gp}{(p+b)}$$

where g is a fixed code book gain factor, b is an adaptive code gain factor and p is a perceptual enhancement gain factor derivable in accordance with;

$$\text{if}\quad\begin{array}{ll} b < TH_{low} & \text{then } p = 0.0 \\ TH_{low} \le b < TH_2 & \text{then } p = a_{enh1} f_1(b) \\ TH_2 \le b < TH_3 & \text{then } p = a_{enh2} f_2(b) \\ \vdots & \vdots \\ TH_{N-1} \le b \le TH_{upper} & \text{then } p = a_{enhN-1} f_{N-1}(b) \\ b > TH_{upper} & \text{then } p = a_{enhN} f_N(b) \end{array}$$

where TH represents threshold values, a_{enh} is a linear scaler and f(b) is a function of gain b.

42. A method for use with Linear Predictive Coding (LPC) for speech synthesis, comprising steps of:

generating first and second excitation signals,

modifying in a LPC decoder the first excitation signal in accordance with a gain factor associated therewith, and

further modifying in the LPC decoder the first excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal in order to produce an enhanced synthesised speech signal.

43. A method for use with Linear Predictive Coding (LPC) for speech synthesis, comprising steps of:

generating first and second excitation signals,

modifying in a LPC decoder the first excitation signal in accordance with a gain factor associated therewith, and

modifying in the LPC decoder the second excitation signal in accordance with a scaling factor derivable from pitch information associated with the first excitation signal in order to produce an enhanced synthesised speech signal.

44. A time domain speech synthesiser, comprising:

an excitation source providing first and second partial excitation signals having a speech periodicity information content; and

a speech quality enhancement post-processor coupled to said excitation source for operating on one of said first and second partial excitation signals, said post-processor modifying the speech periodicity information content of the operated on partial excitation signal in accordance with a signal derivable from at least one of said first and second partial excitation signals.

45. A synthesiser for speech synthesis, comprising:

an input unit for inputting a signal and for extracting coded information from said signal, the coded information comprising fixed codebook and adaptive codebook parameters, including an adaptive codebook gain factor;

an excitation source comprising a fixed codebook and an adaptive codebook and having inputs coupled to outputs of said input unit for receiving extracted coded information therefrom, said excitation source being responsive to the received extracted coded information for outputting a first partial excitation signal from said fixed codebook and a second partial excitation signal from said adaptive codebook, said excitation source further comprising means for combining said first and second partial excitation signals into a composite excitation signal; and

a perceptual enhancement post-processor coupled to said excitation source for operating on said composite excitation signal by combining said composite excitation signal with a scaled version of said second partial excitation signal, wherein an amount of scaling of said second partial excitation signal is controlled by a scaling factor having a value that is function of a value of said adaptive codebook gain factor.

46. A synthesiser as in claim **45**, wherein said input unit inputs said signal from a radio channel.

* * * * *

EXHIBIT H

US006882727B1

(12) **United States Patent**
Vialen et al.

(10) **Patent No.: US 6,882,727 B1**
(45) **Date of Patent: Apr. 19, 2005**

(54) **METHOD OF CIPHERING DATA TRANSMISSION IN A RADIO SYSTEM**

(75) Inventors: **Jukka Vialen**, Espoo (FI); **Fabio Longoni**, Espoo (FI)

(73) Assignee: **Nokia Mobile Phones Ltd.**, Espoo (FI)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/519,184**

(22) Filed: **Mar. 6, 2000**

(30) **Foreign Application Priority Data**

Mar. 8, 1999 (FI) 990500

(51) **Int. Cl.**[7] **H04K 9/08**

(52) **U.S. Cl.** **380/33**; 380/270; 380/259

(58) **Field of Search** 380/259, 33; 280/270

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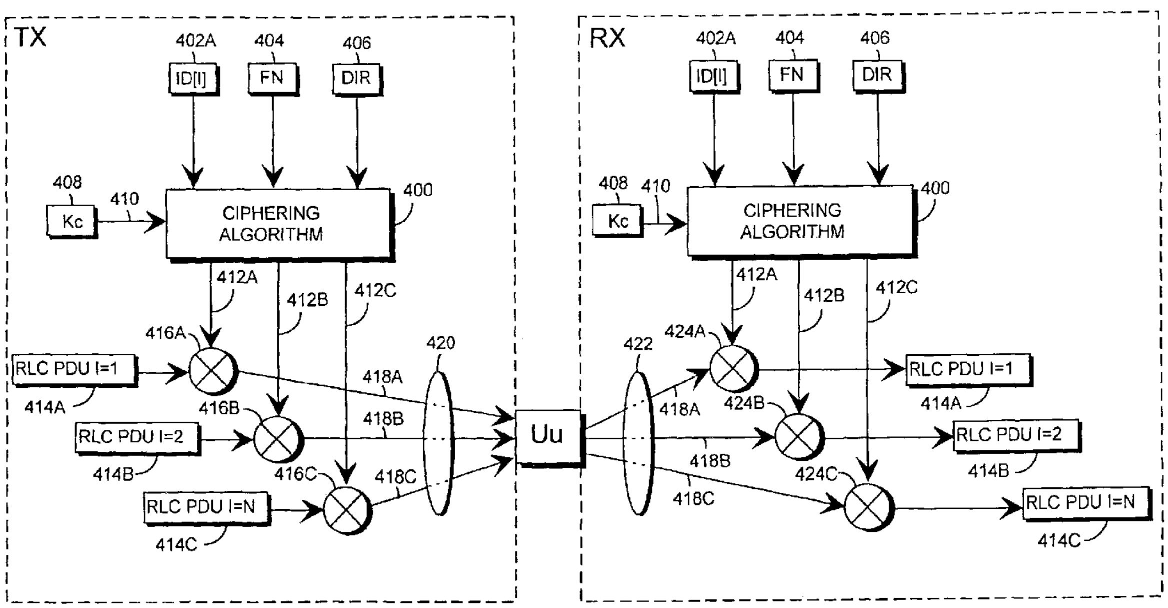
Primary Examiner—Gregory Morse
Assistant Examiner—Ellen Tran
(74) *Attorney, Agent, or Firm*—Perma & Green, LLP

(57) **ABSTRACT**

The invention relates to a method of ciphering data transmission in a radio system, and to a user equipment using the method, and to a radio network subsystem using the method. The method includes the steps of: **(602)** generating a ciphering key; **(604A)** producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter; **(604B)** using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm; and **(606)** producing ciphered data by applying the ciphering mask to plain data.

25 Claims, 13 Drawing Sheets



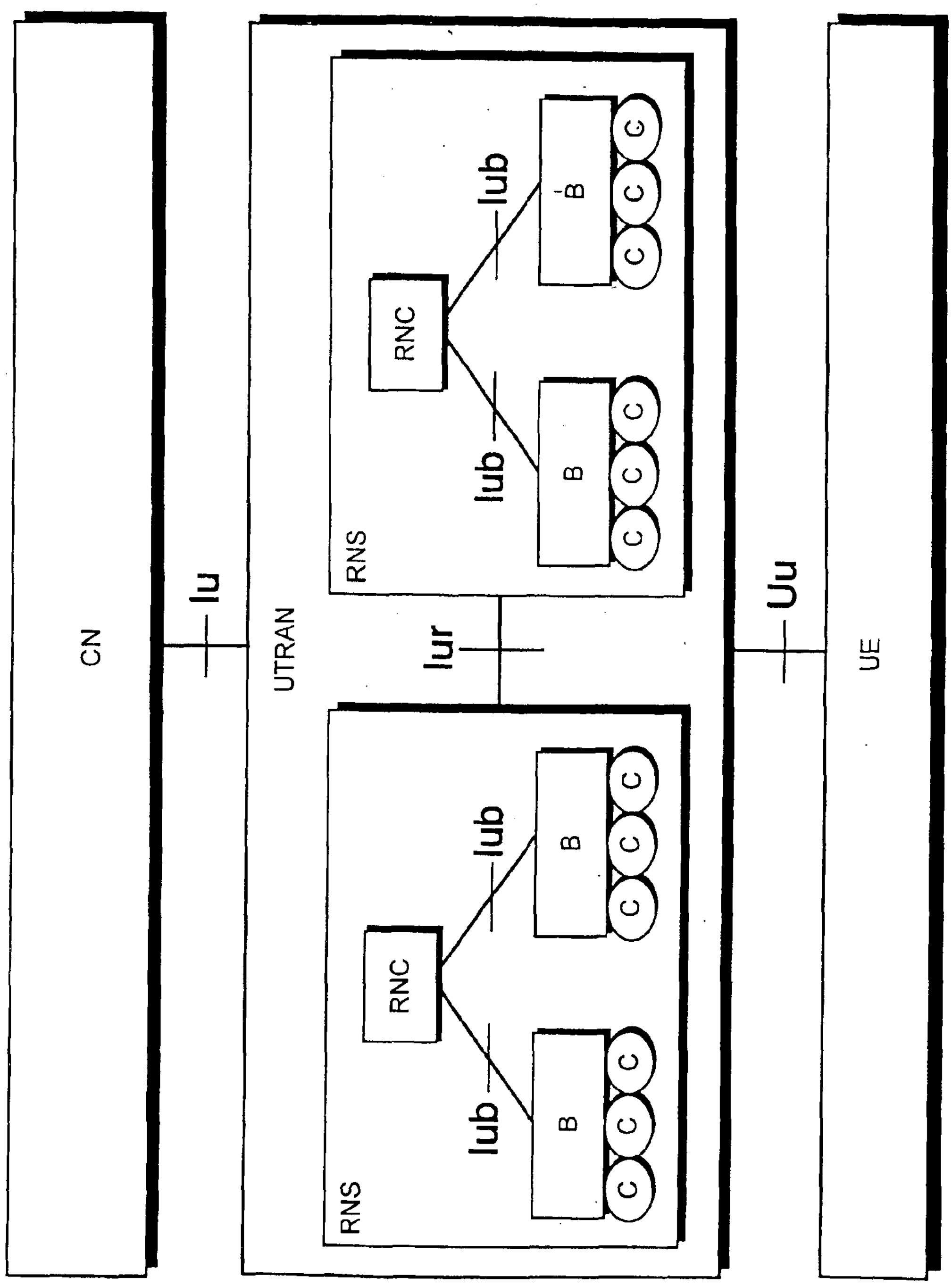
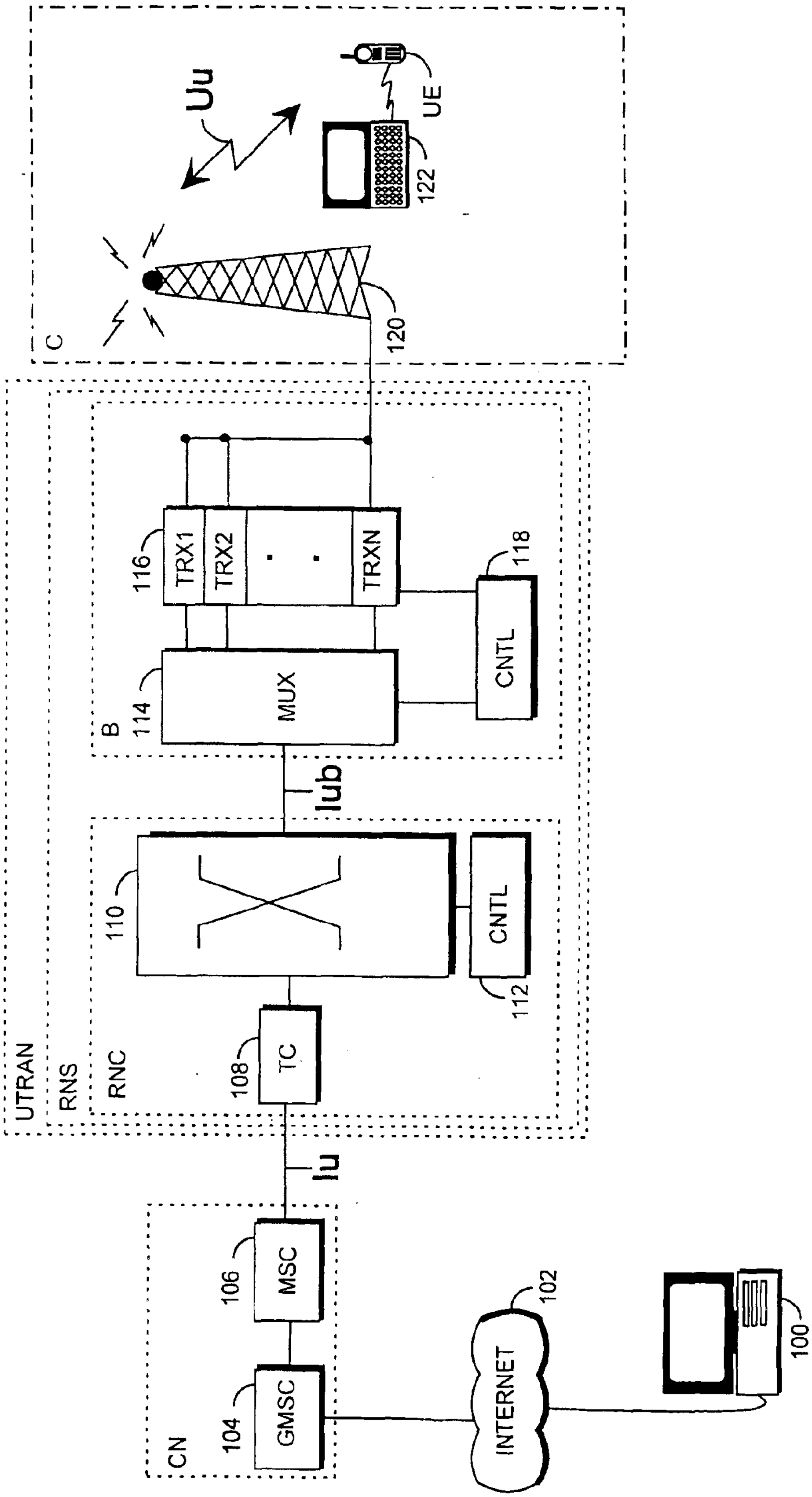


Fig 1A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TRX2
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MSC
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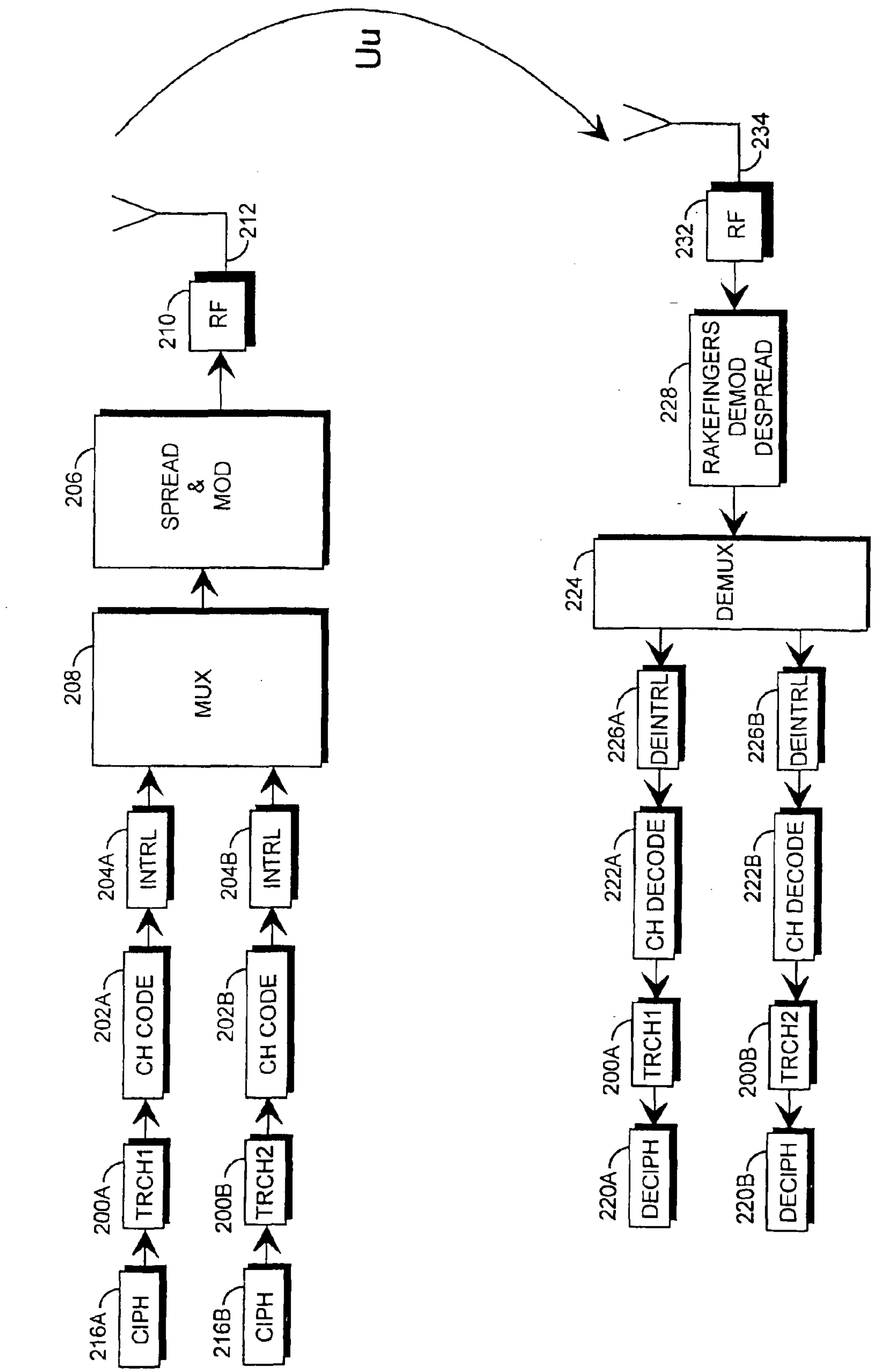
Fig 1B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226B
DEINTRL
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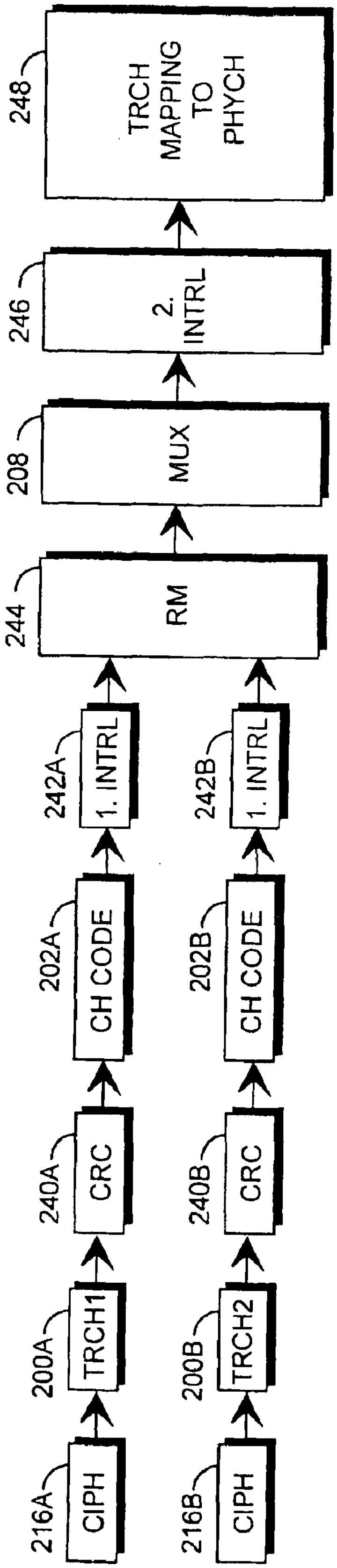
Fig 2A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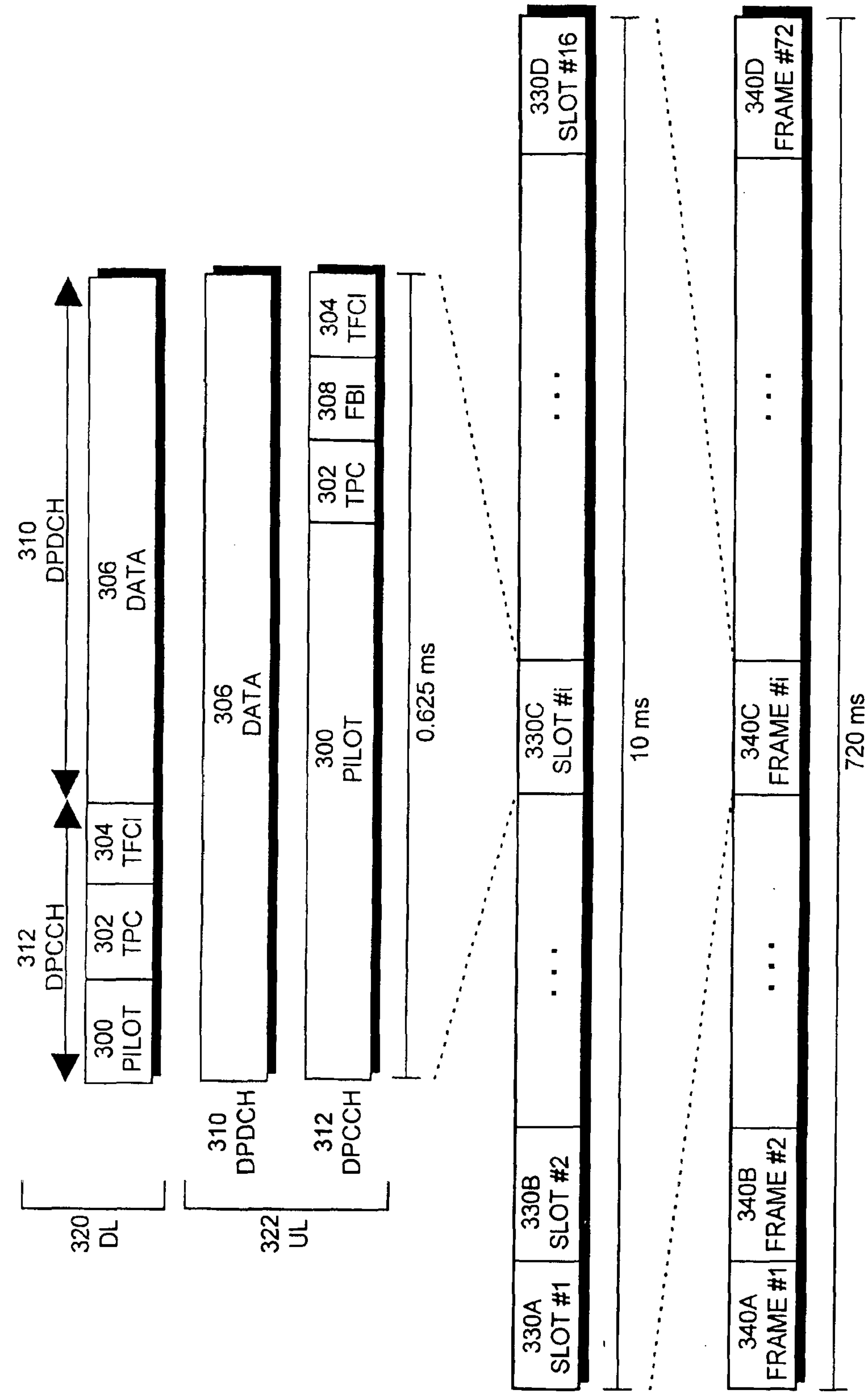
TRCH MAPPING TO PHYCH

Fig 2B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306
DATA
312
DPCCH
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PILOT
302
TPC
308
FBI
304
TFCI
0.625 ms
330A
SLOT #1
330B
SLOT #2
330C
SLOT #i
330D
SLOT #16
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340A
FRAME #1
340B
FRAME #2
340C
FRAME #i
340D
FRAME #72
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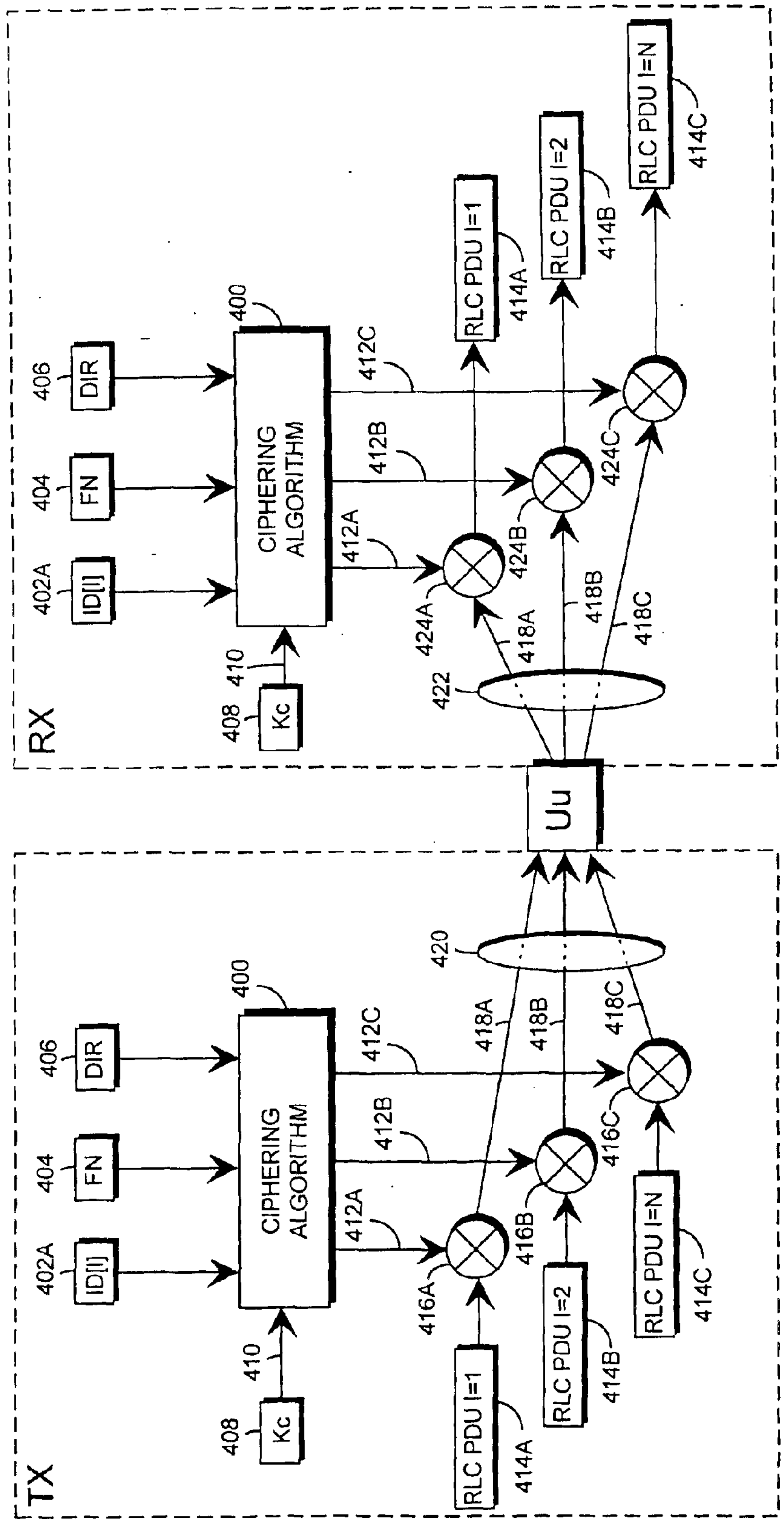
Fig 3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TX
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CIPHERING ALGORITHM
Kc
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FN
DIR
RLC PDU I=1
RLC PDU I=2
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Uu
400
402A
404
406
408
410
412A
412B
412C
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416C
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418B
418C
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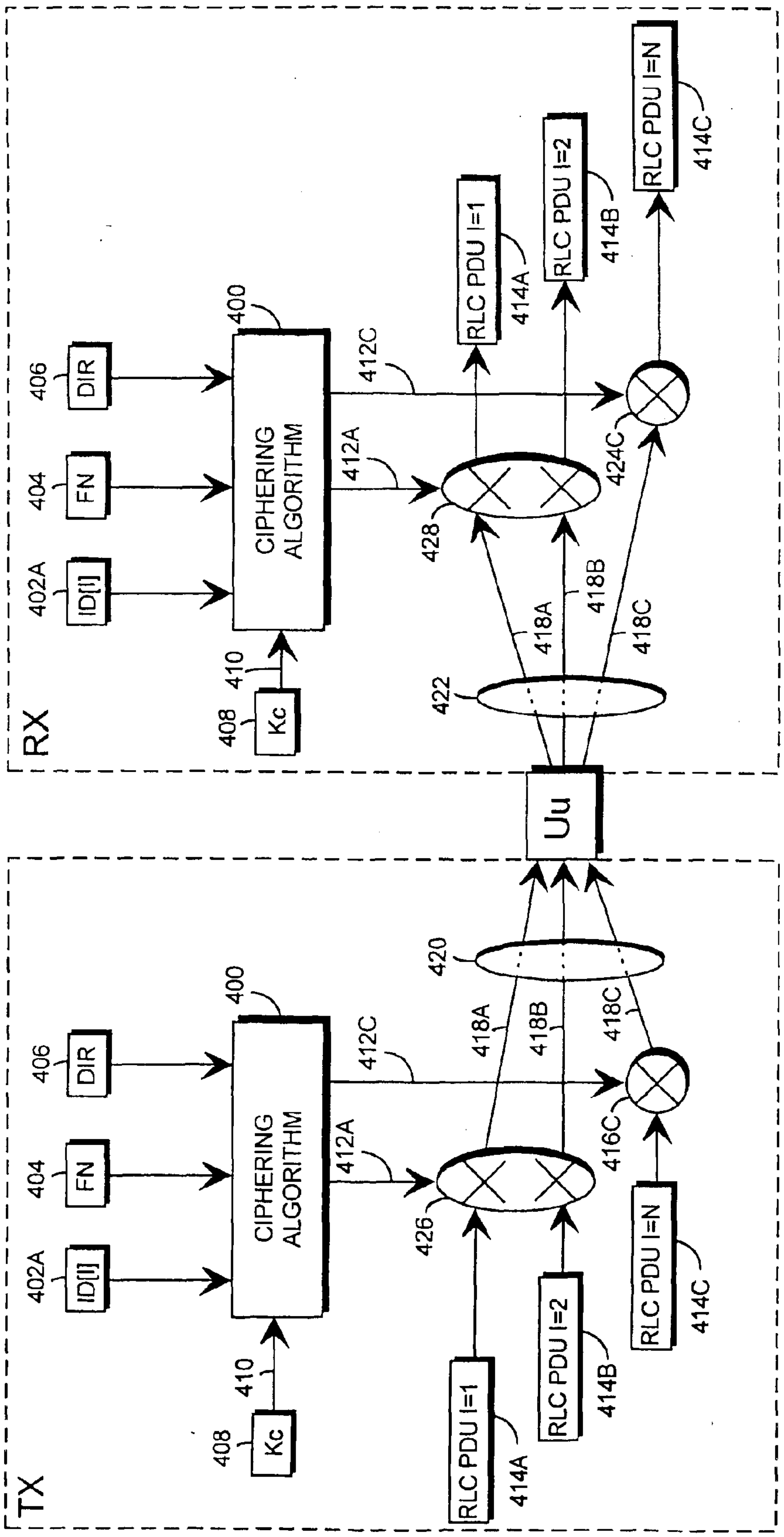
422
424A
424B
424C

Fig 4A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TX
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414A
RLC PDU I=2
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414C
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424C
416C
426
428
Uu

Fig 4B

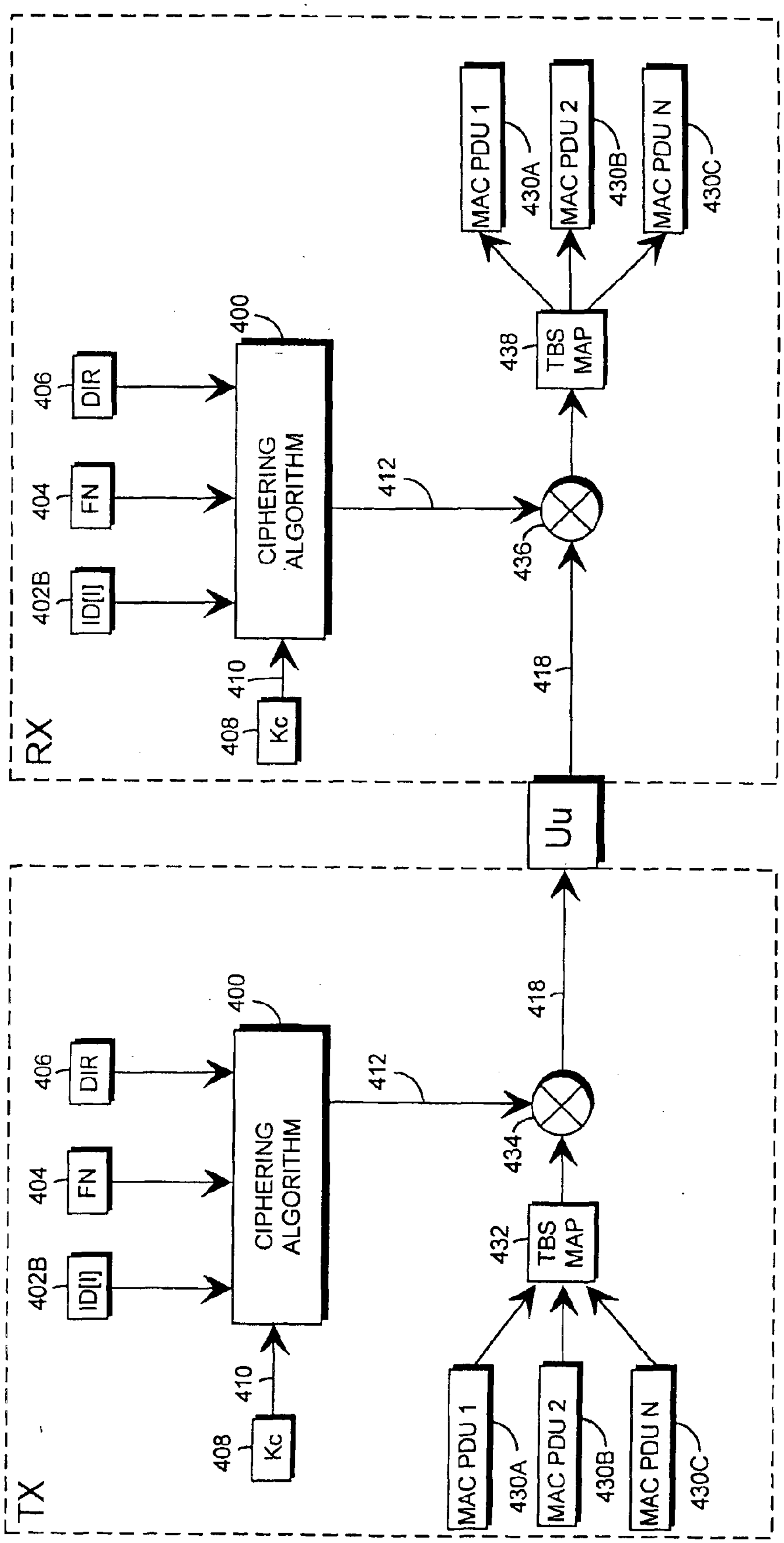
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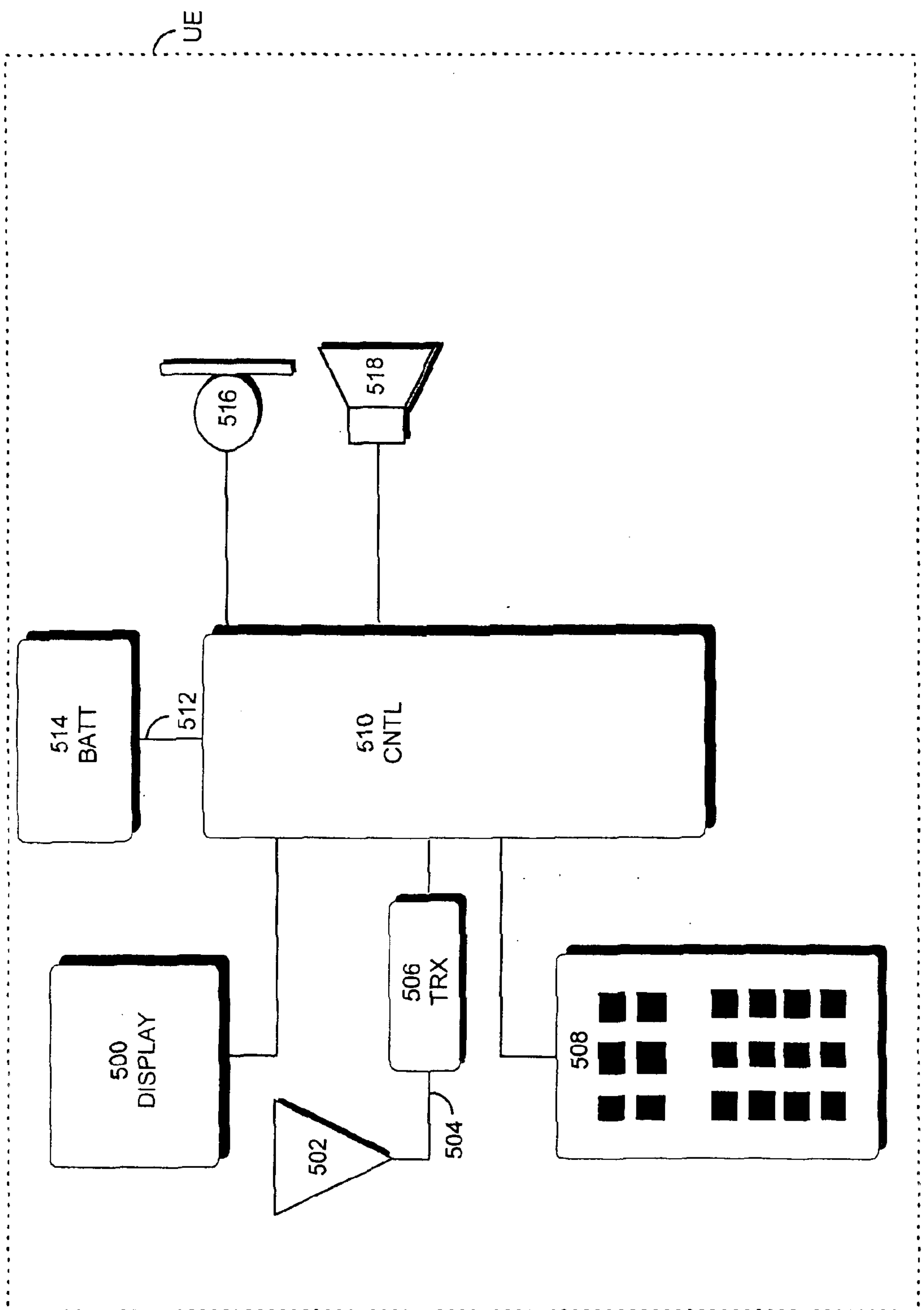
Fig 4C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DISPLAY
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TRX
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504
508

Fig 5

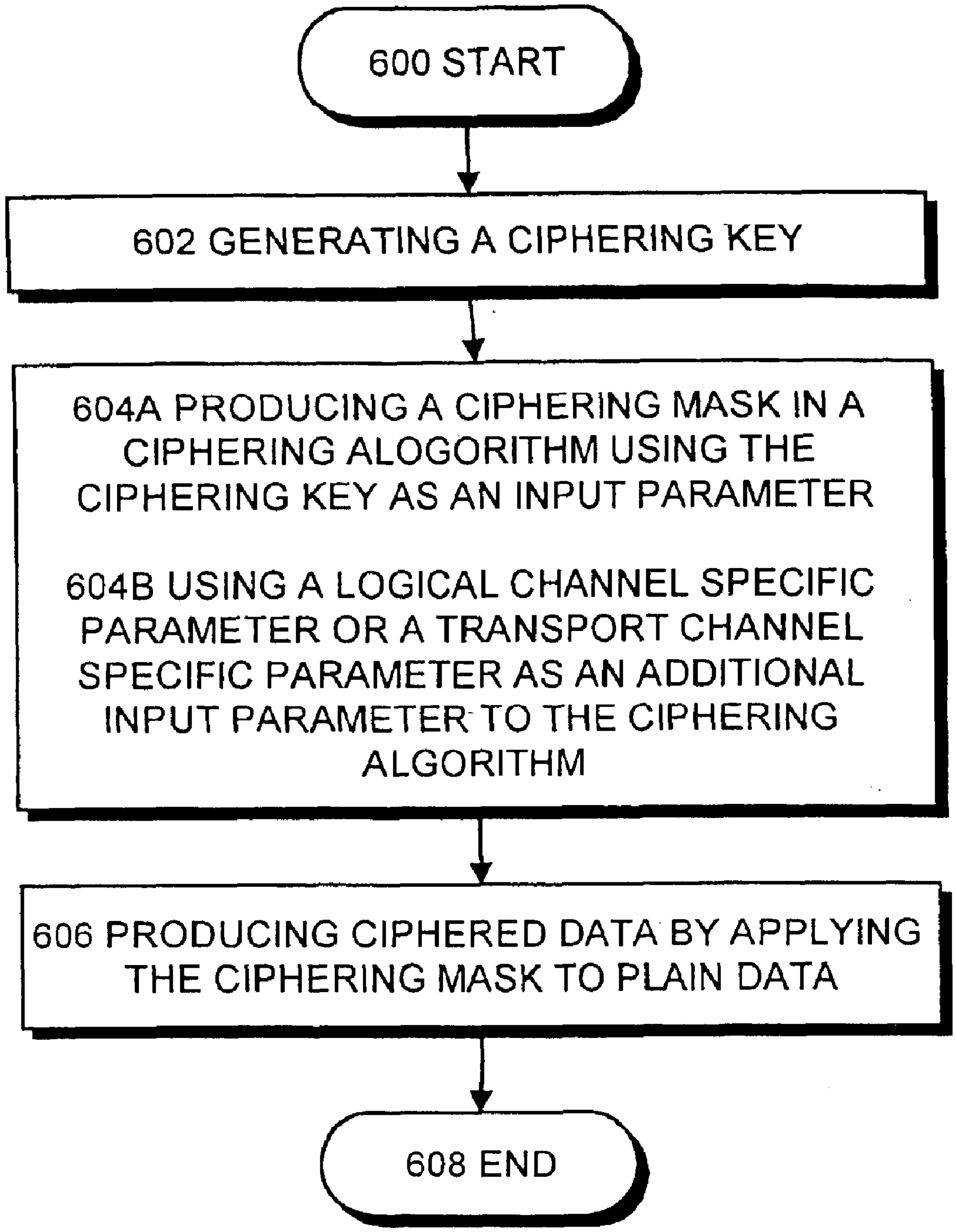
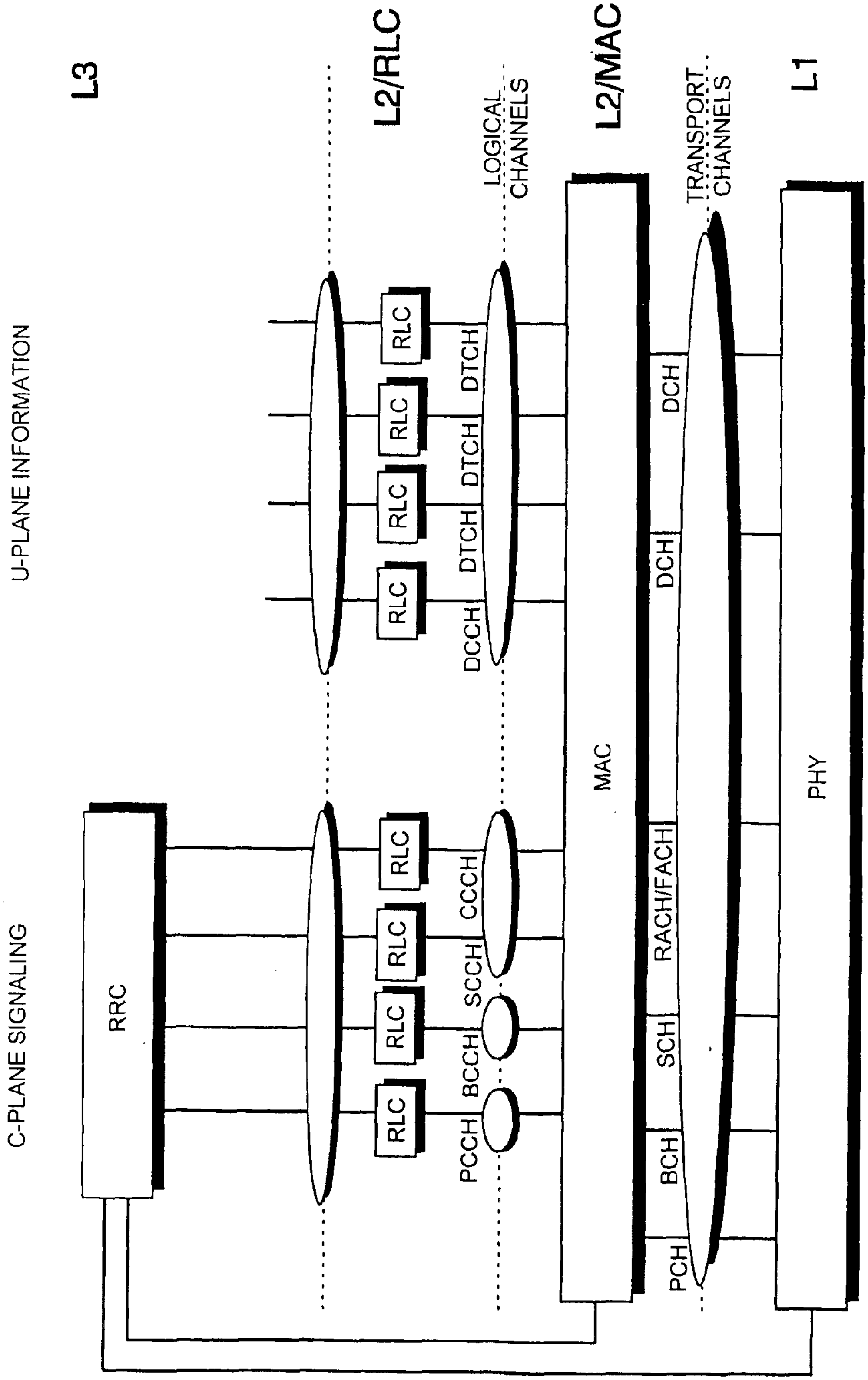

600 START
602 GENERATING A CIPHERING KEY
604A PRODUCING A CIPHERING MASK IN A CIPHERING ALOGORITHM USING THE CIPHERING KEY AS AN INPUT PARAMETER
604B USING A LOGICAL CHANNEL SPECIFIC PARAMETER OR A TRANSPORT CHANNEL SPECIFIC PARAMETER AS AN ADDITIONAL INPUT PARAMETER TO THE CIPHERING ALGORITHM
606 PRODUCING CIPHERED DATA BY APPLYING THE CIPHERING MASK TO PLAIN DATA
608 END

Fig 6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L2/RLC
LOGICAL CHANNELS
L2/MAC
TRANSPORT CHANNELS
L1
U-PLANE INFORMATION
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RLC
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DTCH
DTCH
DTCH
DCCH
CCCH
SCCH
BCCH
PCCH
MAC
DCH
DCH
RACH/FACH
SCH
BCH
PCH
PHY

Fig 7A

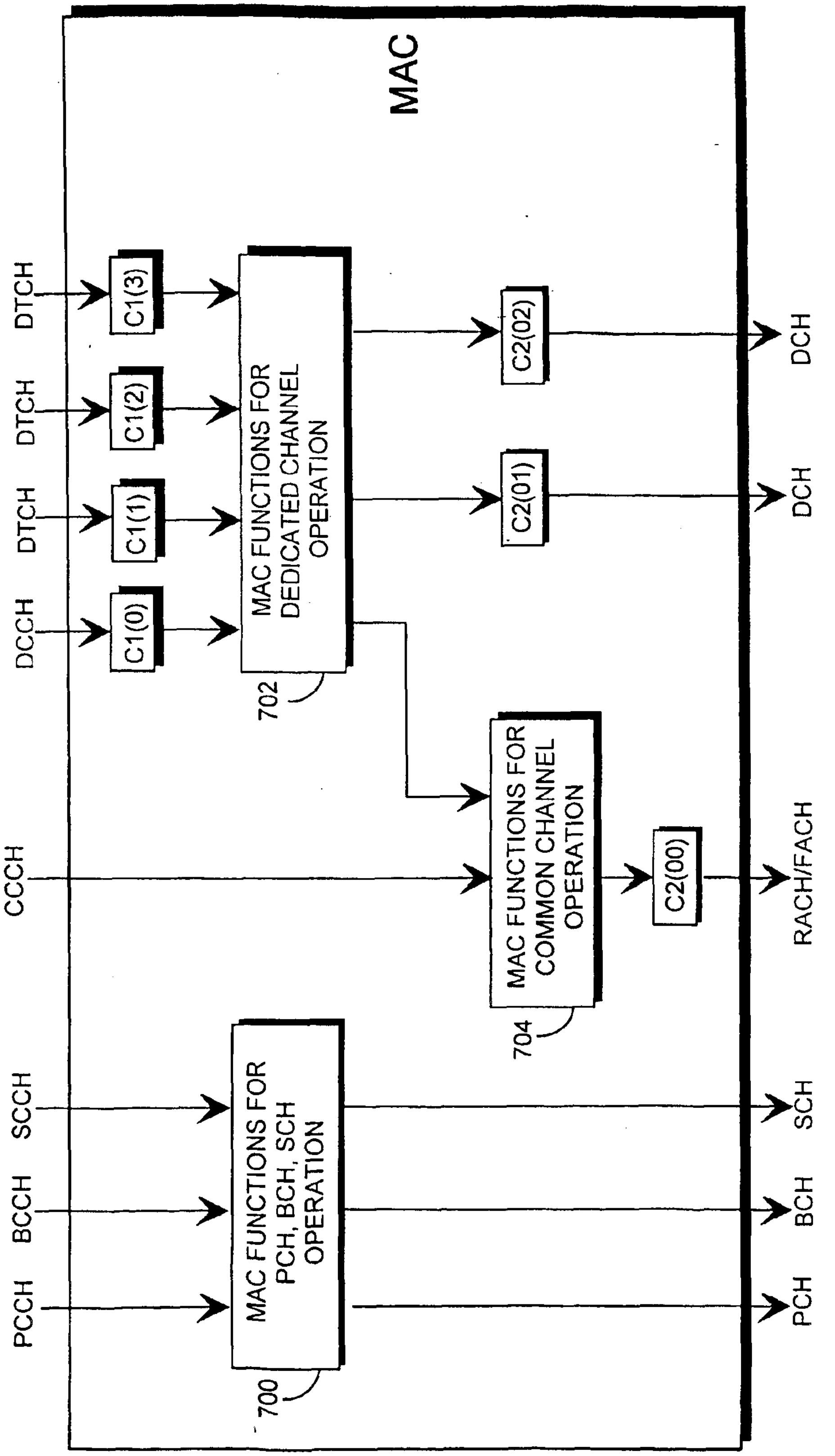
MAC
DCCH
DTCH
DTCH
DTCH
C1(0)
C1(1)
C1(2)
C1(3)
MAC FUNCTIONS FOR DEDICATED CHANNEL OPERATION
702
C2(01)
C2(02)
DCH
DCH
CCCH
MAC FUNCTIONS FOR COMMON CHANNEL OPERATION
704
C2(00)
RACH/FACH
PCCH
BCCH
SCCH
MAC FUNCTIONS FOR PCH, BCH, SCH OPERATION
700
PCH
BCH
SCH

Fig 7B

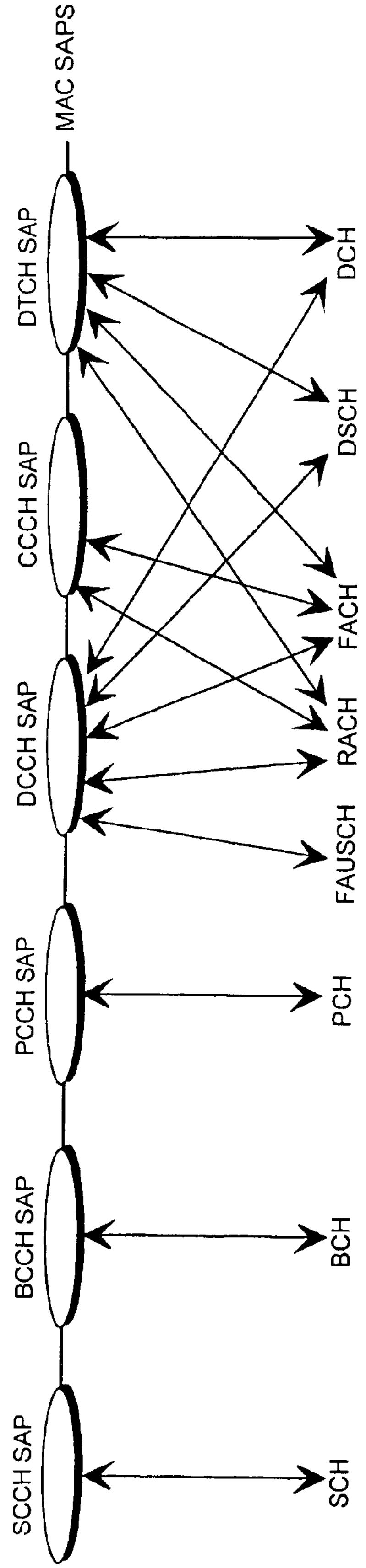
MAC SAPS
DTCH SAP
CCCH SAP
DCCH SAP
PCCH SAP
BCCH SAP
SCCH SAP
DCH
DSCH
FACH
RACH
FAUSCH
PCH
BCH
SCH

Fig 7C

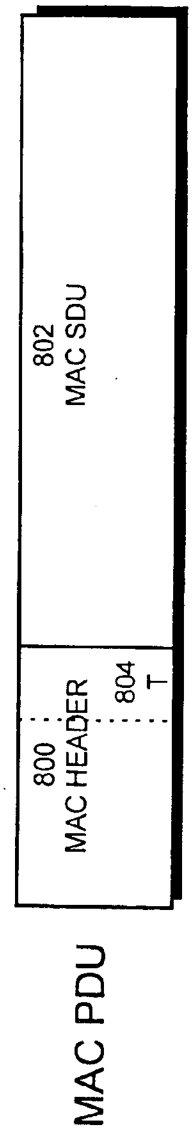
802
MAC SDU
800
MAC HEADER
804
T
MAC PDU

Fig 8

METHOD OF CIPHERING DATA TRANSMISSION IN A RADIO SYSTEM

FIELD OF INVENTION

The invention relates to a method of ciphering data transmission in a radio system.

BACKGROUND OF INVENTION

Ciphering is today used in many data transmission systems to prevent the data transmitted from falling into the hands of an unauthorized user. The ciphering has grown in significance in the past few years, particularly as wireless telecommunication has become more common.

The ciphering can be performed, for example, by encrypting the information to be transmitted in a transmitter, and by decrypting the information in a receiver. In the encryption means the information to be transmitted, for example a bit stream, is multiplied by a certain number of encryption bit patterns, whereby it is difficult to find out what the original bit stream was if the encryption bit pattern used is unknown.

In a digital GSM system, for example, ciphering is performed on the radio path: a ciphered bit stream to be transmitted onto the radio path is formed by XORing data bits with ciphering bits, the ciphering bits being formed by an algorithm known per se (the A5 algorithm), using a ciphering key Kc. The A5 algorithm encrypts the information transmitted on the traffic channel and the DCCH control channel.

The ciphering key Kc is set when the network has authenticated the terminal but the traffic on the channel has not yet been ciphered. In the GSM system the terminal is identified on the basis of the International Mobile Subscriber Identity IMSI, which is stored in the terminal, or the Temporary Mobile Subscriber Identity TMSI, which is formed on the basis of the subscriber identity. A subscriber identification key Ki is also stored in the terminal. A terminal identification key is also known to the system.

In order that the ciphering would be reliable, information on the ciphering key Kc must be kept secret. The cipher key is therefore transmitted from the network to the terminal indirectly. A Random Access Number RAND is formed in the network, and the number is then transmitted to the terminal via the base station system. The ciphering key Kc is formed by a known algorithm (the A5 algorithm) from the random access number RAND and the subscriber identification key Ki. The ciphering key Kc is computed in the same way both in the terminal and in the network part of the system.

In the beginning, data transmission on a connection between the terminal and the base station is thus not ciphered. The ciphering does not start until the base station system sends the terminal a cipher mode command. When the terminal has received the command, it starts to cipher data to be sent and to decipher received data. Correspondingly, the base station system starts to decipher the received data after sending the cipher mode command and to cipher the sent data after the reception and successful decoding of the first ciphered message from the terminal. In the GSM system the cipher mode command comprises a command to start ciphering, and information on the algorithm to be used.

The problem in the known methods is that they have been designed for the present systems, wherefore they are inflexible and not suited for the ciphering of data transmission in new systems, where several parallel services for one mobile station are possible. If we use the same ciphering mask twice for two or more parallel protocol data units that will be sent using the same air interface frame, then an eavesdropper may deduce a lot of information from the data streams. The amount of information that can be deduced depends on the structure of the data streams. From random data that has no structure one cannot obtain any information, but usually there is a structure in the data, especially in the signaling data.

BRIEF DESCRIPTION OF INVENTION

It is an object of the invention to provide a method, and a user equipment and a radio network subsystem implementing the method, solving the above problems. This is achieved with a method of ciphering data transmission in a radio system, comprising: generating a ciphering key; producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter; producing ciphered data by applying the ciphering mask to plain data. Using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm.

The invention also relates to a user equipment, comprising: generating means for generating a ciphering key; a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter; ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data. The ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter.

The invention further relates to a radio network subsystem, comprising: generating means for generating a ciphering key; a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter; ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data. The ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter.

The preferred embodiments of the invention are claimed in the dependent claims.

Several advantages are achieved with the invention. In the solution of the present invention, ciphering and its properties can be flexibly controlled. The present invention enhances user security in new radio systems. This solution is also better than the known technique, which uses a long enough ciphering mask only once for each air interface frame, because it allows distributed implementation of the needed functionality in the protocol stack.

BRIEF DESCRIPTION OF FIGURES

In the following the invention will be described in greater detail by means of preferred embodiments and with reference to the attached drawings, in which

FIGS. **1**A and **1**B illustrate an example of a mobile telephone system;

FIG. **2**A illustrates a transmitter and a receiver;

FIG. **2**B illustrates transport channel coding and multiplexing;

FIG. **3** illustrates a frame structure;

FIGS. **4**A, **4**B and **4**C show a block diagram of a ciphering environment according to the invention;

FIG. **5** illustrates a mobile station

FIG. **6** is a flow diagram illustrating a method according to the invention;

FIG. **7**A illustrates an example of a protocol stack;

FIG. **7**B illustrates an example of a protocol stack according to the invention;

FIG. **7**C illustrates mapping between logical channels and transport channels;

FIG. **8** illustrates the structure of a Medium Access Control Layer Protocol Data Unit.

DETAILED DESCRIPTION OF INVENTION

The present invention can be used in different mobile telephone systems. In the following examples, the use of the invention is described in the Universal Mobile Telephone System (UMTS) without restricting the invention to it. The examples illustrate the FDD (Frequency Division Duplex) operation of the UMTS, but do not restrict the invention to it.

With reference to FIGS. **1**A and **1**B, a typical mobile telephone system structure will be described. FIG. **1**B only comprises the blocks that are essential for the description of the invention, although it is apparent to a person skilled in the art that a common mobile telephone system also comprises other functions and structures, which need not be discussed in greater detail here. The main parts of the mobile telephone system are: a core network CN, a UMTS terrestrial radio access network UTRAN, and a user equipment UE. The interface between the CN and the UTRAN is called the Iu interface, and the interface between the UTRAN and the UE is called the Uu interface.

The UTRAN is composed of radio network subsystems RNS. The interface between two RNSs is called the Iur interface. The RNS is composed of a radio network controller RNC and one or more node Bs B. The interface between the RNC and the node B is called the Iub interface. The reception area of the node B, i.e. cell, is denoted in FIG. **1**A by C.

As the presentation in FIG. **1**A is very abstract, it is clarified in FIG. **1**B by setting forth the parts of the GSM system that correspond to the parts of the UMTS. It is clear that the presented mapping is by no means a binding one but an approximation, because the responsibilities and functions of the parts of the UMTS are still being planned.

FIG. **1**B illustrates a packet switched transmission via Internet **102** from a computer **100** connected with the mobile telephone system to a portable computer **122** connected with a user equipment UE. The user equipment UE may be a fixedly mounted wireless local loop terminal, a vehicle-mounted terminal or a hand-held portable terminal, for example.

The infrastructure of the radio network UTRAN is composed of radio network subsystems RNS, i.e. base station subsystems. The radio network subsystem RNS is composed of a radio network controller RNC, i.e. a base station controller, and at least one node B, i.e. a base station, under the control of the RNC.

The node B comprises a multiplexer **114**, transceivers **116**, and a control unit **118** which controls the operation of the transceivers **116** and the multiplexer **114**. The multiplexer **114** arranges the traffic and control channels used by a plurality of transceivers **116** on a single transmission connection Iub.

The transceivers **116** of the node B have a connection to an antenna unit **120** which is used for providing a

bi-directional (or sometimes one-way) radio connection Uu to a user equipment UE. The structure of the frames transmitted on the radio connection Uu is determined in detail and the connection is referred to as an air interface.

The radio network controller RNC comprises a group switching field **110** and a control unit **112**. The group switching field **110** is used for switching speech and data and for connecting signaling circuits. The node B and the radio network controller RNC form a base station subsystem, which additionally comprises a transcoder, also known as a speech codec, or TRAU (Transcoder and Rate Adapter Unit) **108**.

The division of the functions and the physical structures of the radio network controller RNC and the node B may differ according to the actual realization of the radio network subsystem. Typically, the node B implements the radio connection. The radio network controller RNC typically manages the following: radio resource control, inter-cell handover control, power control, timing and synchronization, and paging for user equipment.

The transcoder **108** is usually located as close to a mobile switching center **106** as possible because this allows speech to be transmitted between the transcoder **108** and the radio network controller RNC in a cellular radio network form, which saves transmission capacity.

The transcoder **108** converts different digital speech coding modes used between a public switched telephone network and a cellular radio network to make them compatible, for instance from the 64 kbit/s fixed network form to another form (such as 13 kbit/s) of the cellular radio network, and vice versa. Naturally, the transcoding is carried out only for speech. The control unit **112** carries out call control, mobility management, collection of statistical data and signaling.

The core network CN is composed of the infrastructure belonging to the mobile telephone system which is not part of the UTRAN. FIG. **1**B illustrates two equipments, which are part of the core network CN, namely a mobile switching center **106**, and a gateway mobile switching center **104**, which handles mobile telephone system interfaces towards the outside world, in this example towards the Internet **102**.

FIG. **5** illustrates an exemplary structure of the user equipment UE. The essential parts of the user equipment UE are: an interface **504** to the antenna **502** of the user equipment UE, a transceiver **506**, a control part **510** of the user equipment UE, an interface **512** to the battery **514**, and a user interface comprising a display **500**, a keyboard **508**, a microphone **516** and a speaker **518**.

FIG. **2**A illustrates the functioning of a radio transmitter/radio receiver pair. The radio transmitter may be located in the node B or in the user equipment. Correspondingly the radio receiver may be located in the user equipment or in the node B.

The upper portion of FIG. **2**A illustrates the essential functionality of the radio transmitter. Different services placed in a physical channel are, for example, speech, data, moving or still video picture, and the control channels of the system that are processed in the control part **214** of the radio transmitter. The control part **214** is related to the control of the equipment itself and to the control of the connection. FIG. **2**A illustrates manipulation of two different transport channels **200**A, **200**B. Different services call for different source encoding equipment: speech for example calls for a speech codec. For the sake of clarity, source encoding equipment is not, however, presented in FIG. **2**A.

First the logical channels are ciphered in blocks **216**A, **216**B. In the ciphering, ciphered data is produced by applying a ciphering mask to plain data. Then the ciphered data is placed in the transport channel in blocks **200**A, **200**B. As later will be explained with reference to FIGS. **4**A, **4**C and **7**B the ciphering can be performed either for a logical channel or for a transport channel. Different channels are then channel encoded in blocks **202**A and **202**B. One form of channel coding is different block codes, one example of which is a cyclic redundancy check, or CRC. Another typical way of performing channel coding is convolutional coding and its different variations, such as punctured convolutional coding and turbo coding.

Having been channel encoded, the channels are interleaved in an interleaver **204**A, **204**B. The object of the interleaving is to make error correction easier. In the interleaving, the bits are mixed with each other in a predetermined fashion, so that transitory fading on the radio path does not necessarily make the transferred information unidentifiable.

Different signals are multiplexed in block **208** so that they can be sent using the same transmitter.

The interleaved encrypted bits are then spread with a spreading code, scrambled with a scrambling code, and modulated in block **206**, whose operation is described in detail in FIG. **2**B.

Finally, the combined signal is conveyed to the radio frequency parts **210**, which may comprise power amplifiers and bandwidth restricting filters. An analog radio signal is then transmitted through an antenna **212** to the radio path Uu.

The lower portion of FIG. **2**A illustrates the typical functionality of a radio receiver. The radio receiver is typically a Rake receiver. The analog radio signal is received from the radio path Uu by an antenna **234**. The received signal is conveyed to radio frequency parts **232**, which comprise a filter that blocks the frequencies outside the desired frequency band. A signal is then converted in a demodulator **228** into an intermediate frequency or directly into baseband, and in this form the signal is sampled and quantized.

Because the signal in question is a multipath propagated signal, efforts are made to combine the signal components propagated on different multipaths in block **228**, which comprises several Rake fingers.

de-interleaved. After that the physical channels are processed in a specific channel decoder **222**A, **222**B, where the channel coding used in the transmission is decoded. Convolutional coding is advantageously decoded with a Viterbi decoder. After this the transport channels are mapped to the logical channels in blocks **200**A, **200**B, or the other possibility is that the deciphering is performed for the transport channels. The channel decoded channels (logical or transport) are deciphered in blocks **220**A, **220**B by applying a ciphering mask to the received data. Each received logical channel can be further processed, for example, by transferring the data to the computer **122** connected with the user equipment UE. The control channels of the system are conveyed to the control unit **236** of the radio receiver.

FIG. **2**B illustrates how the transport channels are coded and multiplexed. In principle, FIG. **2**B is in part the same as FIG. **2**A but seen from another perspective. In blocks **240**A, **240**B a Cyclic Redundancy Check is added to each Transport Block. Interleaving is performed in two stages, in blocks **242**A, **242**B and **246**. When two or more services having different quality of service requirements are multiplexed into one or more physical channels, then service specific rate matching **244** is used. In rate matching the channel symbol rates are adjusted to an optimum level, where the minimum quality of service requirement of each service is fulfilled with the same channel symbol energy. Mapping of the transport channels to physical channels is performed in block **248**.

As the ciphering is the key issue in the current invention, its principle will be next described in more detail. In Table 1 the first row represents the plain data bits that have to be transmitted to the recipient. The bits on the second row constitute a ciphering mask. The ciphering mask is applied to the plain data, usually by using the exclusive-or operation, i.e. XOR. The resulting ciphered data is on the third row. This ciphered data is sent through the air interface to the recipient. The recipient then performs deciphering by applying the same ciphering mask that has been used in the transmitter to the received data. The fourth row is a ciphering mask that is summed with the third row by using the XOR operation. The resulting recovered data is presented on the fifth row. As we will see, the recovered data is the same as the plain data.

TABLE 1

Plain data	0	1	1	1	0	1	0	0	1	1	1	0	0	1	1	1	0	0	0
Ciphering mask	0	0	1	0	1	0	1	0	0	0	1	0	0	0	0	1	1	1	1
Ciphered data	0	1	0	1	1	1	1	0	1	1	0	0	0	1	1	1	1	1	1
Ciphering mask	0	0	1	0	1	0	1	0	0	0	1	0	0	0	0	1	1	1	1
Recovered data	0	1	1	1	0	1	0	0	1	1	1	0	0	1	1	1	0	0	0

In a so-called rowing Rake finger, delays for the different multipath propagated signal components are searched. After the delays have been found, different Rake fingers are allocated for receiving each of the multipath propagated signals by correlating the received signal with the used spreading code delayed with the found delay of that particular multipath. The different demodulated and despread multipaths of the same signal are then combined in order to obtain a stronger signal.

The received physical channel is then demultiplexed in a demultiplexer **224** into data streams of different channels. The channels are then directed each to a de-interleaver **226**A, **226**B, where the received physical channel is then

FIG. **3** shows an example of a frame structure used on a physical channel. Frames **340**A, **340**B, **340**C, **340**D are given a running number from one to seventy-two, and they form a 720-millisecond long super frame. The length of one frame **340**C is ten milliseconds. The frame **340**C is divided into sixteen slots **330**A, **330**B, **330**C, **330**D. The length of slot **330**C is 0.625 milliseconds. One slot **330**C corresponds typically to one power control period, during which the power is adjusted for example by one decibel up or down.

The physical channels are divided into different types, including common physical channels and dedicated physical channels.

The common physical channels are used to carry the following transport channels: PCH, BCH, RACH and FACH.

The dedicated physical channels consist of dedicated physical data channels (DPDCH) **310** and dedicated physical control channels (DPCCH) **312**. The DPDCHs **310** are used to carry data **306** generated in layer two of the OSI (Open Systems Interconnection) model and layers above it, i.e. dedicated control channels (DCH). The DPCCHs **312** carry the control information generated in layer one of the OSI model. Control information comprises: pilot bits **300** used in channel estimation, feedback information (FBI) **308** transmit power-control commands (TPC) **302**, and optionally a transport format combination indicator (TFCI) **304**. The TFCI **304** tells the receiver the transport formats of different transport channels, i.e. Transport Format Combination, used in the current frame.

As can be seen from FIG. **3**, the down-link DPDCHs **310** and DPCCHs **312** are time multiplexed into the same slot **330**C. In the up-link the channels are sent in parallel so that they are IQ/code multiplexed (I=in-phase, Q=quadrature) into each frame **340**C.

The channels in the radio interface Uu are processed according to a protocol architecture comprising, according to the ISO (International Standardization Organization) OSI (Open Systems Interconnection) model, three protocol layers: a physical layer (=layer one), a data link layer (=layer two), and a network layer (=layer three). The protocol stacks are located both in the radio network subsystem RNS and in the user equipment UE. Each unit (e.g. user equipment, or radio network subsystem) has a layer which is in logical communication with a layer of another unit. Only the lowest, physical layers communicate with each other directly. The other layers always use the services offered by the next, lower layer. The message must thus physically pass in the vertical direction between the layers, and only in the lowermost layer the message passes horizontally between the layers. FIG. **7**A illustrates the layers of the protocol architecture. The ovals between different sub-layers indicate service access points (SAP).

The physical layer L**1** offers different transport channels to the MAC sub-layer MAC and higher layers. The physical layer transport services are described by how and with what characteristics data is transferred over the radio interface. The transport channels include a Paging Channel PCH, Broadcast Channel BCH, Synchronization Channel SCH, Random Access Channel RACH, Forward Access Channel FACH, Down-link Shared Channel DSCH, Fast Up-link Signaling Channel FAUSCH, and Dedicated Channel DCH. The physical layer L**1** maps transport channels with physical channels. In the FDD (Frequency Division Duplex) mode a physical channel is characterized by the code, frequency and, in the up-link, the relative phase (I/Q). In the TDD (Time Division Duplex) mode the physical channel is also characterized by the time slot.

The transport channels may be divided into common channels (where there is a need for in-band identification of the UEs when particular UEs are addressed) and dedicated channels (where the UEs are identified by the physical channel, i.e. code and frequency for the FDD and code, time slot and frequency for the TDD).

The common transport channel types are as follows. The RACH is a contention based up-link channel used for transmission of a relatively small amount of data, for example of initial access or non-real-time dedicated control or traffic data. The FACH is a common down-link channel without closed-loop power control used for transmission of a relatively small amount of data. The DSCH is a down-link channel shared by several UEs carrying dedicated control or traffic data. The BCH is a down-link channel used for broadcasting system information to an entire cell. The SCH is a down-link channel used for broadcasting synchronization information to an entire cell in the TDD mode. The PCH is a down-link channel used for broadcasting control information to an entire cell allowing efficient UE sleep mode procedures.

The dedicated transport channel types, in turn, are as follows. The DCH is a channel dedicated to one UE used in up-link or down-link. The FAUSCH is an up-link channel used to allocate dedicated channels in conjunction with the FACH. The data link layer is divided into two sub-layers: a MAC sub-layer (Medium Access Control) and a RLC sub-layer (Radio Link Control). The MAC sub-layer L**2**/MAC offers different logical channels to the RLC sub-layer L**2**/RLC. The logical channel is characterized by the type of information that is transferred. The logical channels include a Paging Control Channel PCCH, Broadcast Control Channel BCCH, Synchronization Control Channel SCCH, Common Control Channel, Dedicated Control Channel DCCH and Dedicated Traffic Channel DTCH.

The control channels are used for transfer of control plane information only. The SCCH is a down-link channel for broadcasting synchronization information in case of TDD (Time Division Duplex) operation. The BCCH is a down-link channel for broadcasting system control information. The PCCH is a down-link channel that transfers paging information. The CCCH is a bi-directional channel for transmitting control information between the network and the UEs. This channel is commonly used by the UEs having no RRC connection with the network. The DCCH is a point-to-point bi-directional channel that transmits dedicated control information between the UE and the network. This channel is established through an RRC connection setup procedure.

The traffic channels are used for the transfer of user plane information only. The DTCH is a point-to-point channel, dedicated to one UE, for the transfer of user information. A DTCH can exist in both up-link and down-link.

The MAC layer maps logical channels with transport channels. One of the functions of the MAC sub-layer is to select the appropriate transport format for each transport channel depending on the momentary source bit rate.

FIG. **7**C illustrates mapping between logical channels and transport channels. An SCCH is connected to an SCH. A BCCH is connected to a BCH. A PCCH is connected to a PCH. A CCCH is connected to a RACH and a FACH. A DTCH can be connected to either a RACH and a FACH, to a RACH and a DSCH, to a DCH and a DSCH, or to a DCH. A DCCH can be connected to either a RACH and a FACH, to a RACH and a DSCH, to a DCH and a DSCH, to a DCH, or to a FAUSCH.

The third layer L**3** has a RRC sub-layer (Radio Resource Control) that handles the control plane signaling of layer three between the user equipment and the network. Among the functions carried out by the RRC sub-layer are assignment, reconfiguration and release of radio resources for the RRC connection. So the RRC sub-layer handles the assignment of the radio resources required for the RRC connection, including the requirements of both the control and the user plane. The RRC layer may reconfigure radio resources during an established RRC connection.

In the present invention we are interested in the encryption of the different services' data flows of one user. According to the known techniques, all data flows would be encrypted using the same ciphering mask.

The method according to the invention for ciphering data transmission in a radio system is presented in FIG. **6**. The performance of the method begins in block **600**.

In block **602** a ciphering key is generated according to a known technique, for example as described in the Background of the Invention section.

In block **604**A a ciphering mask is produced in a ciphering algorithm using the ciphering key as an input parameter. Also a logical channel specific parameter or a transport channel specific parameter is used as an additional input parameter to the ciphering algorithm. The logical channel specific parameter can be one of the following: a Radio Access Bearer Identifier, a Logical Channel Identifier, a Signaling Link Identifier, or some other parameter identifying the logical channel used. The transport channel specific parameter can be, for example, the Dedicated Channel Identifier, or some other parameter identifying the transport channel used.

The term 'bearer' is a high-level name for transmission of information used in connection with a network service. Depending on the services, information in the UMTS can usually be transmitted using one or more bearers. The services include, for example, speech transmission, data services and video service. A radio bearer, on the other hand, represents that part of the bearer which extends over the air interface. One logical channel normally carries one radio bearer. A logical channel defines the service offered by the MAC layer. A logical channel can be mapped to different types of transport channels depending on the existing service mode (either to a dedicated transport channel or common transport channels). The transport channels define the services offered by the physical layer. It is also possible to multiplex several logical channels into one transport channel in the MAC layer. The transport channels are further mapped to physical channels in the physical layer. Several transport channels can be multiplexed into one physical channel by layer **1**. It is also possible that after transport channel multiplexing the data stream is divided between several physical channels.

The invention can thus be applied to a radio system whose terminals can communicate with other transceivers using one or more parallel radio bearers. Typically, when a call is established between a terminal and a network, a physical channel is first established for a Signaling Radio Bearer SRB between the terminal and the radio network subsystem, and once this channel has been established, the actual traffic bearer(s) can be established. The SRB can also be called a signaling link.

The direction of transmission (up-link/down-link) can be used as an additional input parameter to the ciphering algorithm.

Yet another parameter exists: a radio frame specific parameter can be used as an additional input parameter to the ciphering algorithm. The radio frame specific parameter can be, for example, the User Equipment Frame Number (UEFN), or some other parameter identifying the used radio frame. The radio frame specific parameter depends on the protocol layer where the ciphering function is implemented. If it is implemented in the protocol layer that is terminated in the UE and the CN, then a mechanism for conveying the used frame number to the receiving entity has to be defined. If the ciphering function is located in the MAC layer or layer **1** (or some other layer terminated in the UE and the node B or the RNC), a frame number at least partly consisting of the

physical frame number can be used, which means that the used frame number need not be signaled with the data.

In block **606** ciphered data is produced by applying the ciphering mask to plain data, using for example the XOR operation as described in Table 1.

Next, an elaborated example illustrating the implementation of the ciphering method in the transmitter and in the receiver is explained in connection with FIGS. **4**A, **4**B and **4**C. Only the relevant points will be illustrated, but it will be clear for a person skilled in the art how ciphering can be performed in various situations for example with different numbers of PDUs.

FIG. **4**A describes a block diagram defining the basic ciphering environment defined in this invention. Generating means **408** are used for generating a ciphering key **410** according to a known technique. Connected with the generating means **408** there is a ciphering algorithm **400** for producing ciphering masks **412**A, **412**B, **412**C. The ciphering algorithm uses the generated ciphering key **410** as an input parameter. The ciphering algorithm **400** uses a logical channel specific parameter **402**A as an additional input parameter.

In the receiver end, the logical channel specific parameter needed for deciphering can be read from an unciphered MAC header, for example from the C/T-field of the MAC header. The structure of the MAC PDU is illustrated in FIG. **8**. The MAC PDU consists of an optional MAC header **800** and a MAC Service Data Unit (MAC SDU) **802**. Both the MAC header and the MAC SDU are of variable size. The content and the size of the MAC header **800** depend on the type of the logical channel, and in some cases none of the parameters in the MAC header **800** are needed. The size of the MAC-SDU **802** depends on the size of the RLC PDU, which is defined during the set-up procedure. The MAC header **800** comprises a C/T-field **804**. This option allows efficient MAC multiplexing of different logical channels (or different instances of the same logical channel type) into one transport channel, both into dedicated transport channels and common transport channels. When this method is used, the MAC header is not ciphered, which allows separating the different MAC PDUs in the receiver end and which in the common channel mode allows reading the RNTI (Radio Network Temporary Identity) field that is needed for routing messages to the correct entity in the UTRAN.

Connected with the ciphering algorithm **400** there are ciphering means **416**A, **416**B, **416**C for producing ciphered data **418**A, **418**B, **418**C by applying the ciphering mask **412**A, **412**B, **412**C to the plain data **414**A, **414**B, **414**C. As can be seen from FIG. **4**A, the plain data includes Radio Link Control Layer Protocol Data Units from at least two parallel logical channels, and for each logical channel an individual ciphering mask is produced. So in FIG. **4**A the ciphering masks **412**A, **412**B and **412**C are all different from each other.

In block **420** the ciphered RLC-PDUs are processed through the MAC layer and mapped into one Transport Block Set, i.e. MAC PDU Set.

Another possible solution is one in which the plain data includes one Radio Link Control Layer Protocol Data Unit **414**A from only one logical channel, and for said logical channel an individual ciphering mask **412**A is produced. So the invention also works for the individual logical channel.

Normally a new ciphering mask is produced for each radio frame of the physical layer of the protocol stack. If interleaving is used, then a new ciphering mask can be produced for each interleaving period of the physical layer of the protocol stack. Typically one interleaving period consists of several radio frames.

The left-hand side of FIG. **4**A represents the operations carried out in the transmitter. The corresponding operations will also be carried out in the receiver, as illustrated on the right-hand side of FIG. **4**A. The only differences are that block **422** is used to derive RLC-PDUs out of the received Transport Block Set, and that the deciphering means **424**A, **424**B, **424**C are used to decipher the received data.

In one embodiment of the invention, a Radio Link Control Layer Protocol Data Unit of at least one logical channel is already ciphered, and the step of producing ciphered data is not repeated for said already ciphered Radio Link Control Layer Protocol Data Unit. It is thus avoided that the data would be ciphered twice. Of course, if for example such end-to-end ciphering is used, the data can be ciphered twice: first by the application using the service, and then by the MAC layer according to the invention. This will cause no loss of transmission capacity, as the XOR operation does not add any extra bits, even if it is performed twice.

FIG. **4**B illustrates a solution to a situation where the plain data includes at least two successive Radio Link Control Layer Protocol Data Units of one logical channel. If we assume, for example, that the first RLC PDU **414**A and the second RLC PDU **414**B are from one logical channel, then the problem can be solved in such a way that only one ciphering mask **412**A is produced for these PDUs **414**A, **414**B. Different parts of this ciphering mask **412**A are then used for ciphering the first PDU **414**A and the second PDU **414**B. The length of the required ciphering mask **412**A in this case is naturally the sum of the lengths of the first and the second PDU **414**A, **414**B. Because the PDUs **414**A, **414**B are from the same logical channel (same Radio Access Bearer), the maximum length required can be calculated as being two times the maximum RLC PDU size of that bearer.

FIG. **4**C illustrates a situation where the plain data includes one Transport Block Set (TBS) including Medium Access Control Layer Protocol Data Units of at least two different logical channels, and for each Transport Block Set one ciphering mask **412** is used in producing the ciphered data. In this option, the basic unit to be ciphered is a Transport Block Set. This defines the required length of the ciphering mask **412** produced by the algorithm **400**. Layer **1** still adds Transport Block specific CRCs (Cyclic Redundancy Check), but because the XOR operation does not change the length of data, it should be possible to cipher the whole TBS as one unit. The length of each transport block in the TBS has to be told to L**1** anyway. This option has the disadvantage that the MAC header is also ciphered, and so the MAC PDUs cannot be routed anywhere on the network side before the TBS is deciphered. This is a problem if common channels over Iur are possible. The length of the required ciphering mask **412** is equal to the maximum Transport Block Set size for the transport channel in question.

Another possible solution is one in which the plain data includes one Transport Block Set including a Medium Access Control Layer Protocol Data Unit of one logical channel, and for each Transport Block Set one ciphering mask is used in producing the ciphered data.

The solution of the invention is implemented in the radio system preferably by software, whereby the invention requires certain functions in the protocol processing software located in the transmitter and in the receiver, especially in blocks **204**A, **204**B and **226**A, **226**B of FIG. **2**A. Thus the generating means **408**, the ciphering algorithm **400**, and the ciphering means **416**A, **416**B, **416**C can be software modules of the protocol stack residing in the user equipment UE and in the radio network subsystem RNS. The solution can also be implemented with hardware, for example using ASIC (Application Specific Integrated Circuit) or discrete components.

The method of the invention can be implemented, for example, in the Medium Access Control Layer of the protocol stack. This is illustrated in FIG. **7**B, which shows a high-level overview of the MAC layer depicted in FIG. **7**A with ciphering functions included. C**1**( ) and C**2**( ) are two alternatives for the location of ciphering. C**1**(**0**), C**1**(**1**), C**1**(**2**) and C**1**(**3**) refer to the use of logical channel specific ciphering parameters as explained above with reference to FIGS. **4**A and **4**B, whereas C**2**(**00**), C**2**(**01**) and C**2**(**02**) refer to the use of transport channel specific ciphering parameters. Some MAC functions may be needed below C**2**(**00**), C**2**(**01**) and C**2**(**02**) blocks, but for the sake of clarity they are not illustrated here. Basically the RLC PDUs come to the MAC layer from each logical channel. In the MAC layer the RLC-PDUs are then mapped to the MAC PDUs in the functional blocks **700**, **702**, **704**, which include the operations for the PCH, BCH, SCH, Dedicated Channel and Common Channel operations. Normally one RLC PDU is mapped to one MAC PDU (=Transport Block). This mapping realizes the mapping from a logical channel to a transport channel. The mapping rules have been explained above in connection with FIG. **7**C. If ciphering is used for the CCCH then a ciphering block, for example C**1**(**4**), should be in FIG. **7**B in the line between the 'CCCH' and the functional block **704**.

Even though the invention is described above with reference to an example shown in the attached drawings, it is apparent that the invention is not restricted to it, but can vary in many ways within the inventive idea disclosed in the attached claims.

What is claimed is:

1. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphering data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm,

wherein the logical channel specific parameter is one of the following: a Radio Access Bearer Identifier, a Logical Channel Identifier, a Signaling Link Identifier.

2. The method as claimed in claim **1**, further comprising:

using the direction of transmission as an additional input parameter to the ciphering algorithm.

3. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm,

wherein the transport channel specific parameter is a Dedicated Channel Identifier.

4. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm;

using a radio frame specific parameter as an additional input parameter to the ciphering algorithm;

wherein the radio frame specific parameter is a User Equipment Frame Number.

5. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm,

wherein the plain data includes Radio Link Control Layer Protocol Data Units from at least two parallel logical channels, and for each logical channel an individual ciphering mask is produced.

6. The method as claimed in claim **5**, wherein a Radio Link Control Layer Protocol Data Unit of at least one logical channel is already ciphered, and the step of producing ciphered data is not repeated for said already ciphered Radio Link Control Layer Protocol Data Unit.

7. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm,

wherein the plain data includes at least two successive Radio Link Control Layer Protocol Data Units of one logical channel, and for each Radio Link Control Layer Protocol Data Unit a different part of the ciphering mask is used in producing the ciphered data.

8. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter;

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific parameter or a transport channel specific parameter as an additional input parameter to the ciphering algorithm,

wherein the plain data includes one Transport Block Set including Medium Access Control Layer Protocol Data Units of at least two different logical channels, and for each Transport Block Set one ciphering mask is used in producing the ciphered data.

9. A method of ciphering data transmission in a radio system, comprising:

generating a ciphering key;

producing a ciphering mask in a ciphering algorithm using the ciphering key as an input parameter,

producing ciphered data by applying the ciphering mask to plain data;

using a logical channel specific paramter or a transport channel specific parpmeter as an additional input parameter to the ciphering algorithm,

wherein the plain data includes one Transport Block Set including a Medium Access Control Layer Protocol Data Unit of one logical channel, and for each Transport Block Set one ciphering mask is used in producing the ciphered data.

10. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or transport channel specific parameter as an additional input parameter,

wherein the logical channel specific parameter is one of the following: a Radio Access Bearer Identifier, a Logical Channel Identifier, a Signaling Link Identifier.

11. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the transport channel specific parameter is a Dedicated Channel Identifier.

12. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter;

wherein the ciphering algorithm uses a radio frame specific parameter as an additional input parameter, and

the radio frame specific parameter is a User Equipment Frame Number.

13. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transpoirt channel specific parameter as an additional input parameter;

wherein the ciphering means accept plain data including Radio Link Control Layer Protocol Data Units from at least two parallel logical channels, and the ciphering algorithm produces for each logical channel an individual ciphering mask, and the ciphering means use for each logical channel the ciphering mask of said channel.

14. The user equipment as claimed in claim **13**, wherein a Radio Link Control Layer Protocol Data Unit of at least one logical channel is already ciphered, and the ciphering means do not cipher said already ciphered Radio Link Control Layer Protocol Data Unit.

15. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including at least two successive Radio Link Control Layer Protocol Data Units on one logical channel, and the ciphering algorithm produces for said logical channel an individual ciphering mask, and the ciphering means use for each Radio Link Control Layer Protocol Data Unit different part of the ciphering mask.

16. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including one Transport Block Set including Medium Access Control Layer Protocol Data Units of at least two different logical channels, and the ciphering algorithm produces for each Transport Block Set an individual ciphering mask, and the ciphering means use for each Transport Block Set one ciphering mask.

17. A user equipment, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including one Transport Block Set including a Medium Access Control Layer Protocol Data Unit on one logical channel, and the ciphering algorithm produces for each Transport Block Set an individual ciphering mask, and the ciphering means use for each Transport Block Set one ciphering mask.

18. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter;

wherein the logical channel specific parameter is one of the following: a Radio Access Bearer Identifier, a Logical Channel Identifier, a Signaling Link Identifier.

19. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the transport channel specific parameter is a Dedicated Channel Identifier.

20. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering algorithm uses a radio frame specific parameter as an additional input parameter, and

the radio frame specific parameter is a User Equipment Frame Number.

21. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional parameter,

wherein the ciphering means accept plain data including Radio Link Control Layer Protocol Data Units from at least two parallel logical channels, and the ciphering algorithm produces for each logical channel an individual ciphering mask, and the ciphering means use for each logical channel the ciphering mask of said channel.

22. The radio network subsystem as claimed in claim **21**, wherein a Radio Link Control Layer Protocol Data Unit of at least one logical channel is already ciphered, and the ciphering means do not cipher said already ciphered Radio Link Control Layer Protocol Data Unit.

23. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key an an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including at least two successive Radio Link Control Layer Protocol Data Units of one logical channel, and the ciphering algorithm produces for said logical channel an individual ciphering mask, and the ciphering means use for each Radio Link Control Layer Protocol Data Unit a different part of the ciphering mask.

24. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including one Transport Block Set including Medium Access Control Layer Protocol Data Units of at least two different logical channels, and the ciphering algorithm produces for each Transport Block Set an individual ciphering mask, and the ciphering means use for each Transport Block Set one ciphering mask.

25. A radio network subsystem, comprising:

generating means for generating a ciphering key;

a ciphering algorithm connected with the generating means for producing a ciphering mask using the ciphering key as an input parameter;

ciphering means connected with the ciphering algorithm for producing ciphered data by applying the ciphering mask to plain data;

the ciphering algorithm uses a logical channel specific parameter or a transport channel specific parameter as an additional input parameter,

wherein the ciphering means accept plain data including one Transport Block Set including a Medium Access Control Layer Protocol Data Unit of one logical channel, and the ciphering algorithm produces for each Transport Block Set an individual ciphering mask, and the ciphering means use for each Transport Block set one ciphering mask.

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