Radio Access Schemes and Technologies for Next-Generation Network

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Multimedia services such as Web browsing, video on demand, and IP telephone delivered to the office and home by wired network systems such as optical fiber systems and ADSL systems are now very popular. This popularity increases the demand for more powerful technologies to provide radio access such as high data rate, QoS controlling. The 3rd generation wireless system already has the ability to provide such multimedia services, but advancement of radio access is necessary for stress-free operation. In this paper, we summarize the requirements for next-generation radio access schemes to be put in place around 2010. A high data rate is the most important requirement for multimedia services. Orthogonal frequency division multiplexing (OFDM) in combination with multiple input multiple output (MIMO) multiplexing is considered to be the method most likely to satisfy this requirement. Here, we introduce some advanced technologies that help improve the transmission data rate and the spectral efficiency for the next-generation wireless systems.

1. Introduction

The requirements for next-generation broadband radio access are increasing due to the advance of technologies for next-generation wired network systems. **Figure 1** shows the evolution of radio access. The 1st generation systems were analog and could not provide data access. The 2nd generation systems, which were launched around 1995, had digital technologies and could work with data access. However, the data transmission rate of these systems was not sufficient to provide multimedia services. The 3rd generation systems were launched around 2000 to provide multimedia services and are expected to provide the services shown in **Figure 2**. These systems used to employ circuit switching for voice transmission. However, packet-based switching with better efficiency is expected for multimedia services, and all-IP operation is one of the most important features for next-generation wireless networks.

There are various standardization groups for next-generation radio access. Two of them that are well known are the Long Term Evolution (LTE) of the 3rd Generation Partnership Project (3GPP) and 802.16 (WiMAX) of the IEEE. The International Telecommunication Union/Study Group 8 (ITU/R SG8) is also discussing next-generation radio access and has named the next-generation system "IMT-Advanced." The World Radio Communication Conference (WRC-07) will be held in October 2007 to discuss the new frequency bands for IMT-Advanced.

Section 2 of this paper summarizes the requirements for next-generation radio access, and Section 3 introduces the new technologies that will satisfy these requirements. Section 4 concludes the paper.

2. Requirements for next-generation radio access

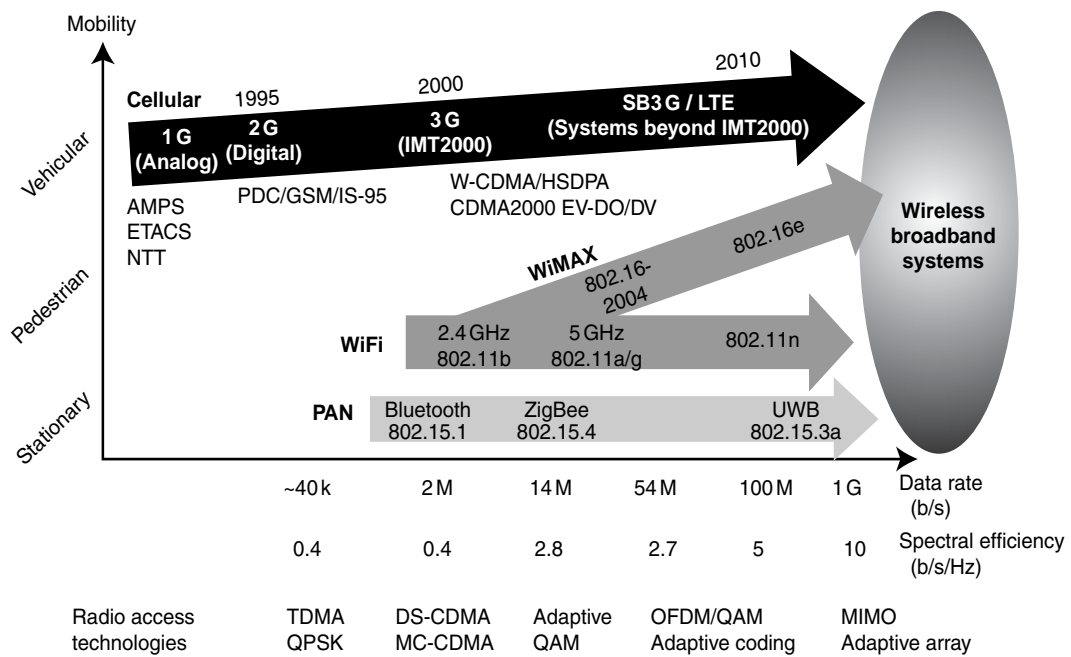Next-generation radio access is expected to
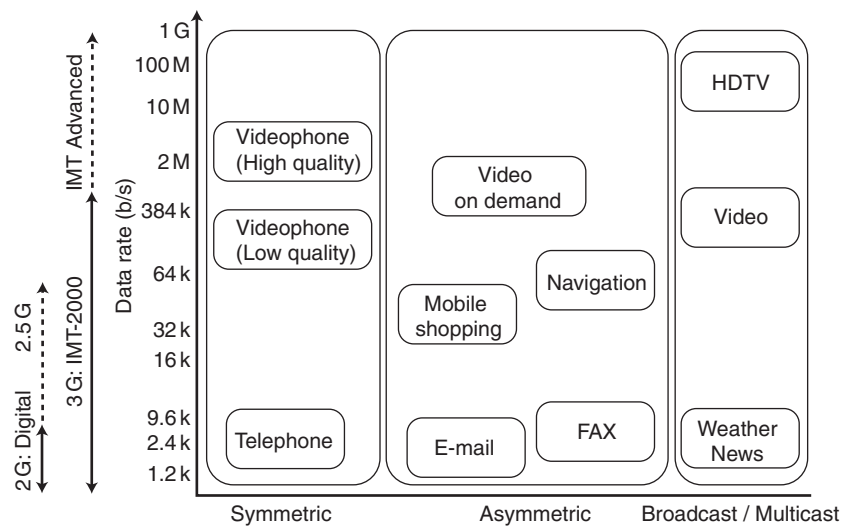
Figure 1
Mobile multimedia services.

Figure 2
Mobile multimedia services.

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provide a 1 Gb/s or less data rate under stationary conditions and about a 100 Mb/s data rate under vehicular conditions.[1] Figure 1 shows three types of evolutions for the next generation. A personal area network (PAN) is used for short-distance, stationary communications. The other types are candidates for middle-distance, mobile communications. Wide frequency band operation and high spectral efficiency are needed for data transmission rates over 100 Mb/s. Low latency and flexible operations are also needed for multimedia services. Furthermore, all users should be able to easily receive services under various conditions.

We summarize the requirements for next-generation radio access below and describe some techniques for satisfying these requirements.

1) High peak data rate operation
- Wide frequency band operation
  – Orthogonal frequency division multiplexing (OFDM)
- Improvement of spectral efficiency (more than 5 b/s/Hz)
  – Multiple input multiple output (MIMO) multiplexing
  – Higher-order modulation
- Improvement of data rate at the cell-edge
  – Low-rate channel coding
  – Interference coordination/cancellation
  – Transmitter beam-forming/adaptive array antenna reception
2) Multimedia operations
- Realize low-delay and highly reliable radio transmission using error control techniques.
  – Hybrid automatic repeat request (HARQ)
- Enable flexible allocation of radio resources according to the required transmission rate and QoS
  – Orthogonal frequency division multiple access (OFDMA)
  – Frequency and time domain scheduling
3) Operation conditions
- Support a maximum terminal speed of 100 km/h (preferably, a maximum of approx-

imately 300 km/h)
– Advanced channel estimation

3. Air interface and individual technologies for next generation

3.1 Outline of air interface (OFDMA)

Before we explain the individual technologies, we will outline the air interface of the next-generation wireless systems. The OFDM-based access scheme is considered the most promising candidate for next-generation wireless systems. OFDM is a multi-carrier parallel transmission method in which the carrier frequency is divided into orthogonal sub-carriers. To eliminate the effect of inter-symbol interference caused by multi-path delay spread, a guard period called the "cyclic prefix" is inserted between OFDM symbols. Because OFDM is tolerant of multi-path interference, a high peak data rate can be achieved using higher order modulations such as 16 QAM and 64 QAM, which improves the spectral efficiency of the system.

Furthermore, OFDMA is widely believed to be the most promising candidate for next-generation radio access. As shown in **Figure 3**, total bandwidth is divided into sub-channels. Each sub-channel consists of several sub-carriers and is allocated to the user who has the best channel quality on that sub-channel. Generally, the base
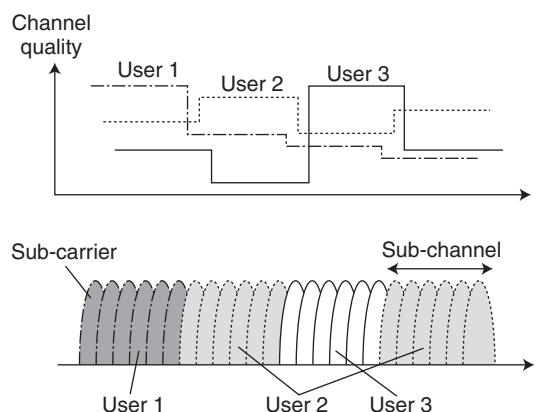

Figure 3
Principle of OFDMA.

station executes the frequency and time domain scheduling and dynamically allocates sub-channels to the users. OFDMA improves the average channel quality between the base station and mobile terminals, which in turn improves total cell throughput. OFDMA has been adopted by the IEEE as standard 802.16e (mobile WiMAX) and is also under discussion at the 3GPP LTE as one of the most promising candidates for the new radio access scheme.

3.2 Individual technologies
1) MIMO

The transmission data rate for next-generation radio access can be improved by applying OFDM-based technology, which is tolerant of multi-path interference. However, the application of higher-order modulation is not sufficient to achieve the required peak data rates, which range from 100 Mb/s to 1 Gb/s. Therefore, it is thought that MIMO is indispensable for satisfying such a high-level requirement. **Figure 4** shows the basic structure of MIMO multiplexing, in which multiple antennas transmit separate data to increase the data rate. At the receiver, signal processing is executed to demultiplex and recover the transmitted data using the estimated channel information of the MIMO propagation channel.[2]

There are MIMO transmission techniques other than MIMO multiplexing, for example, space time block coding (STBC), in which data symbols are encoded into the transmissions of each antenna, and transmitter beam-forming, which controls the transmission weight of each transmitter antenna element. While MIMO multiplexing increases the maximum data rate at high signal noise ratio (SNR) region, STBC and transmitter beam-forming improve the reception performance at low SNR region. In a mobile environment, the received signal strength and the multi-path angle distribution at the transmitter and receiver vary according to the motion of the terminal. Therefore, the total system throughput is expected to improve if the system could dynamically switch between MIMO schemes according to the propagation environment. However, switching between MIMO schemes requires changes in both the transmission schemes and reception algorithms, which increases the complexity of the signal processing and the overall scheme. To resolve this problem, we propose a novel MIMO transmission scheme that uses multiple fixed beams[3] (**Figure 5**). In the proposed multi-beam MIMO scheme, multiple beams are used at the base station to transmit multiple data-streams. Beams and data-streams are adaptively selected at mobile stations, and the selected information
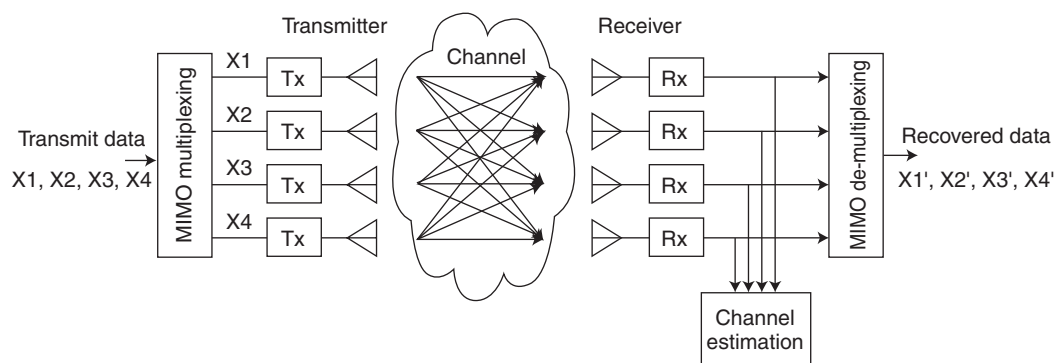

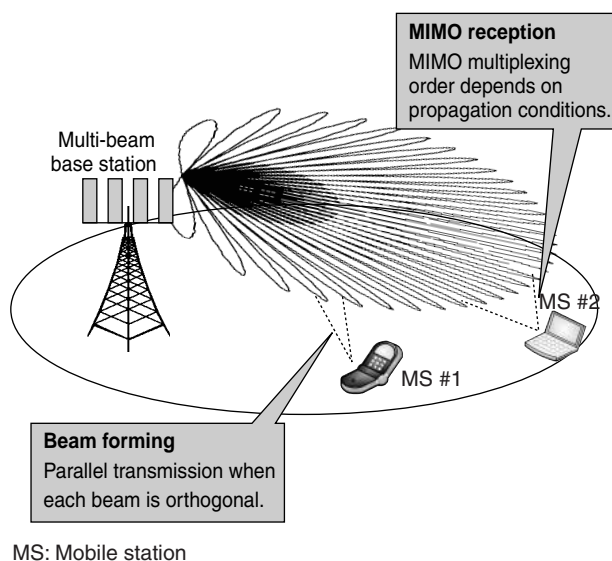Figure 4
Basic structure of MIMO multiplexing.

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MIMO reception
MIMO multiplexing order depends on propagation conditions.

Multi-beam base station

MS #2

MS #1

Beam forming
Parallel transmission when each beam is orthogonal.

MS: Mobile station

Figure 5
Principle of multi-beam MIMO transmission.

Transmitted signal

Received signal

Signal strength

Direct path

Delayed path

Time

Figure 6
Time variance.

is fed back to the base station.[4] MIMO multiplexing is applied if more than two beams are selected, and transmitter beam-forming is applied if only one beam is selected. Therefore, multibeam MIMO by itself can cover the entire cell by switching smoothly between the two MIMO schemes; namely, MIMO multiplexing and transmitter beam-forming.

2) Channel estimation method

The time variance of channel conditions such as signal strength, phase shift, and path profile is an important issue for radio communication systems. **Figure 6** shows an example of time variance. Signals transmitted on various radio propagation paths are combined at the receiver. In this figure, there is a direct path and a delayed path, and the delay and signal strength change according to the movement of the receiver. Time variance of channel conditions is especially high in urban areas and when the receiver is moving fast. The receiver needs to understand this variance in order to obtain the correct data stream, and this is achieved using a function that we call "channel estimation." Channel estimation requires a reduction of operation steps and accurate estimation results. These requirements
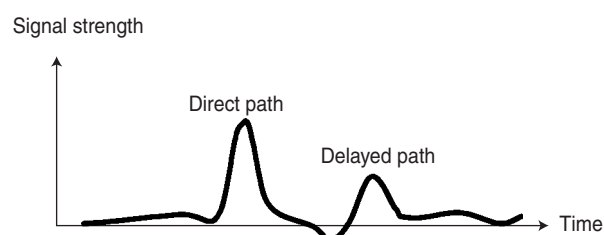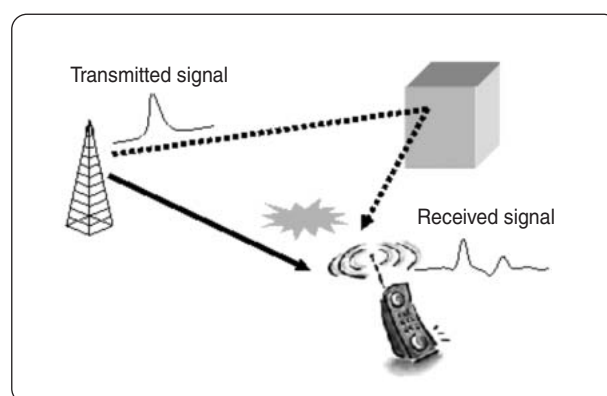
conflict with one another, so we propose a solution to compensate for the conflict called degenerated inverse matrix-based channel estimation (DIME).[5]

The principle of DIME is shown below. The receiver obtains a perfect channel condition by using a cyclic sinc-function matrix that is uniquely determined by each transmitted subcarrier. Because this sinc-function ("time response of a subcarrier" in a broad sense) is a deterministic and known vector, the inverse matrix approach can be used for high-precision estimation without supplementary information such as the channel statistics and operating SNR. A reduction of operation steps is also possible. Some elements of the matrix are set to zero when the received signal in the time domain is below the presented threshold, and then the degenerated sub-matrix is obtained.

A simulation of the bit error rate (BER) for DIME indicated that it almost reaches the theoretical limit (**Figure 7**). We also performed a

laboratory experiment that indicated a broadband communication of over 100 Mb/s at terminal speeds of 300 km/h can be achieved using DIME.

3) Channel coding scheme

In a cellular system, it is important that communication quality be excellent throughout the coverage area. However, communication quality deteriorates when the receiver is far from the base station. That is, the signal is degraded by propagation loss and interference from neighboring base stations. A low-rate channel coding is introduced to overcome this problem. This technology can achieve better communication quality than simple repetition coding because of its coding gain.

We examined whether we could achieve a coding gain by applying "shortened code," which is a form of low-rate channel coding, to Turbo code.[6),7)] Because shortened code can be realized with a small modification to the conventional Turbo encoder/decoder, we proposed this scheme as an alternative to simple repetition coding at the 3GPP LTE.[8)] **Figure 8** shows the procedure for making shortened codes. Suppose that the length of a transmitter's information sequence is
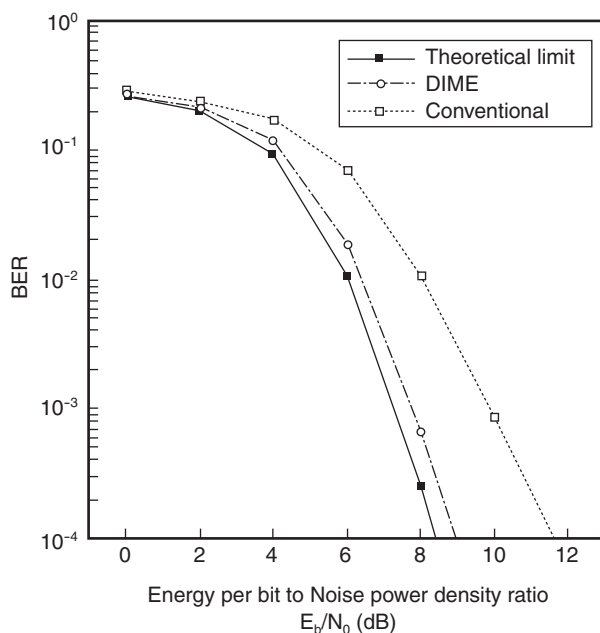
k bits, the length of the temporary bit sequence is $k0$, and the total bit sequence is encoded by a mother code whose coding rate is $R0$. As a result, m parity bits are added, and the coded bit sequence becomes $n0$. If the mother code is a systematic code, the temporary bits can be deleted from the coded bits to raise the efficiency of the transmission. The shortened code is transmitted through the channel. At the receiver, we can insert $k0$ size temporary bits with maximum reliability and recover the correct information.

The code rate R in this scheme is as follows:

$$R0 = \frac{k+k0}{n0}, \quad R = \frac{k}{n} = \frac{k}{n0-k0}, \tag{1}$$

where n is the length of the shortened code.

Figure 9 shows the block diagram of a Turbo encoder that uses shortened code Type 1, which has the merit of being implementable on conventional encoders and decoders. However, the performance of Type 1 is not very good because the positions of the temporary bits are changed by the interleaver before they enter the second encoder. To improve the performance, the positions can be uniformly distributed; however, this does not guarantee the integrity of the insertion pattern after the interleaver.

To solve this problem, we directly inserted temporary bits before each encoder. This is shown in **Figure 10**. This method is called Type 2 and has a better performance than Type 1.
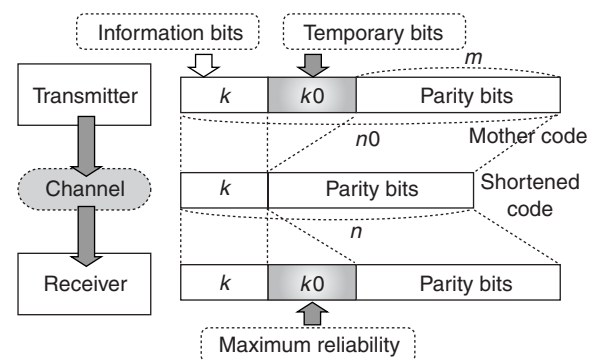
Figure 11 shows the results of a simulation

Figure 7
BER simulation results.

Figure 8
Procedure for making shortened codes.
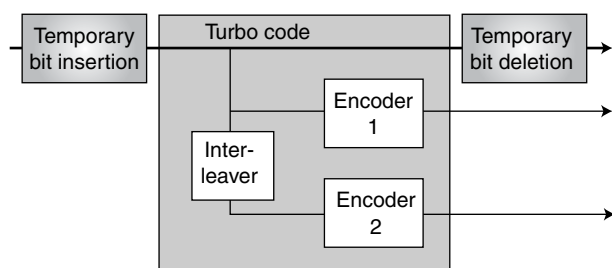
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Figure 9
Block diagram of shortened code Type 1.
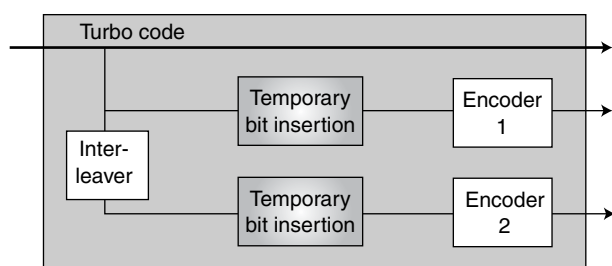
Figure 10
Block diagram of shortened code Type 2.

Figure 11
Required E_b/N_0 vs. coding rate at BLER of 0.1 under AWGN conditions.

of the required E_b/N_0 versus the coding rate using Type 1 and Type 2 shortened Turbo code at a block error rate (BLER) of 10^{-1} under additive white Gaussian noise (AWGN) conditions. The results indicate a performance gain of about 0.2 to 0.25 dB for Type 1 and 0.2 to 0.4 dB for Type 2 compared to the simple repetition technique[9] at a coding rate of 1/4 to 1/7.

4. Conclusion

Radio access is an important element for the next-generation wireless networks. In this paper, we summarized the requirements for next-generation radio access and described examples of an air interface for radio access and individual techniques for satisfying these requirements. Multimedia operation is the most important requirement for next-generation systems. Therefore, a high data rate and high-quality transmission are urgently needed. In this paper, we discussed a part of our research for improving the data rate and communication quality.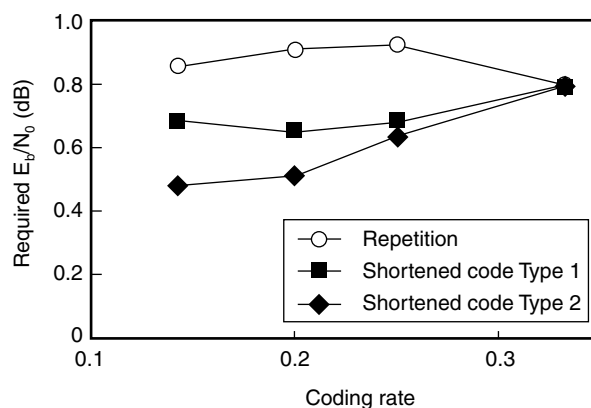 Further study about how wired and wire-less networks will work together and about systematic improvement will be needed.

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