Abstract

A number of factors have motivated most spectrum owners to make plans to deploy uphanded second-generation cellular technologies for use in the PCS bands in the United States. However, the second-generation cellular technologies will need to be enhanced to provide third-generation services. There is interest in using the PCS and cellular spectra to broaden the market of users and the range of use of wireless beyond where it stands today, and beyond the primary applications that drove the development of the technologies being deployed. There is also interest in new technologies to improve the performance of cellular and PCS services, reduce the cost, and improve availability. This article considers evolutionary changes to IS-136 TDMA to enable it to provide a variety of PCS concepts. These evolutionary changes are presented in the form of options that would 1) provide high-quality voice service for indoor and pedestrian systems such as cellular office systems and personal base stations; 2) support enhanced-bit-rate packet wireless data access to the Internet as well as circuit data access; 3) provide smart antenna technology to improve coverage, quality, and capacity; 4) automatically assign frequencies for operation and provide for dynamic channel reconfiguration; 5) support microcellular arrangements to provide low-cost and high-capacity service in dense areas; and 6) support a future high-speed packet data access mode through a wideband system that is complementary to IS-136 TDMA and supports single-terminal operation.

The Evolution of IS-136 TDMA for Third-Generation Wireless Services

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This article considers evolutionary changes to IS-136 time-division multiple access (TDMA) to enable it to provide a variety of personal communications services (PCS) concepts: higher voice quality for indoor and pedestrian modes, enhanced packet and circuit wireless data, smart antenna technology to improve coverage, quality, and capacity, dynamic channel assignment to balance traffic, support for microcellular to provide low-cost and high-capacity service in dense areas, and support for a future high-speed packet data access mode through a wideband system complementing IS-136. Figure 1 depicts these evolutionary changes and options for IS-136 TDMA. A complementary future high-speed packet data mode with bit rates up to 384 kbps will require the introduction of wider channels, but it will be possible to support both services based on 30 kHz channels and high-speed packet data channels in a single terminal. In the following sections, each of these evolutionary changes is discussed. The next section discusses enhanced voice services. This includes a subsection on speech coding options and another on modulation, channel coding, and diversity strategies needed to support the higher bit rate needed for enhanced voice. We then discuss enhanced speed for IS-136 data services. The article examines uplink and downlink smart antenna technologies for IS-136 services, and discusses automatic and dynamic channel assignment for IS-136. We discuss a microcellular evolutionary path for IS-136, and very briefly discuss how IS-136 could be combined with a complementary wideband high-speed packet data wireless access scheme. Each of these proposals is evolutionary in nature. When considered in total, they show how IS-136 can evolve to reach the vision of wireless PCS illustrated in Fig. 1.

Enhanced Voice Services

Cellular office systems, indoor wireless systems, and microcellular systems have been the focus of a number of technology and market trials and service introductions in recent years [1-8]. These trials have generally shown promising results for in-building microcellular systems, particularly when these systems were integrated with conventional macrocellular systems to provide seamless access anywhere. The increased ability to be reached for subscribers with a single-handset/single-number capability can result in increased productivity for the subscriber and higher satisfaction with wireless services. Furthermore, these trials have demonstrated a willingness to pay for indoor wireless access that is reasonably priced. About 88,000 handsets were shipped in the United States for in-building systems in 1995 with an equipment revenue of about $100 million, and the growth rate was 134 percent compared to 1994. This market is expected to reach 1.6 million units at $1.2 billion by 2000 [9]. In Europe, there is increasing interest in combining Global System for Mobile Communications (GSM) or digital cellular services at 1800 MHz (DCS-1800) technology with Digital European Cordless Telephone (DECT) technology in a single multimode handset [10]. Standards are being developed to support network interoperation between GSM-based cellular systems and privately owned DECT-based keysets or private branch exchanges (PBXs). Ericsson and others are developing dual-mode GSM/DECT handsets [11]. A number of service operators are planning to deploy GSM-based PCS technology in the United States, and are likely to make dual-mode GSM/DECT-based technologies available there if this approach is successful in Europe. In Asia, there is interest in dual-mode GSM/personal handphone service (PHS) or personal digital cellular (PDC)/PHS handsets. This approach potentially supports wireline-quality speech with low transmission delay in private indoor environments while permitting a single personal terminal to support wireless access for both indoor private wireless access and widespread cellular access. A disadvantage of this approach is the need for a handset to support two air interfaces with little similarity.
This approach also requires separate spectra for private and public systems.

As depicted in Fig. 1, the IS-136 TMA standard offers the opportunity for integrating macrocellular and indoor wireless access in a seamless fashion while using a low-cost handset based entirely on 30 kHz frequency-division duplex (FDD) channels. The IS-136 standard includes provisions for private system IDs, and the integration of private and public wireless access. Since IS-136 is based on the narrow-band 30 kHz channels with hundreds of channels in a commercial system, schemes that borrow channels from a public system for the operation of a low-power private indoor system can work well. Private systems can borrow spectrum in small pieces, and generally can find quiet spectrum while operating within a macrocellular system. Studies of personal base stations (PBSs) operating within a macrocellular system using shared spectrum have shown that user densities as high as 300 to 1250 PBSs per square mile can be supported [12, 13].

The requirement for high-quality speech is likely to be most pronounced for the macrocellular, residential cordless, wireless keyset, and wireless PBX environments where the speech quality will be compared directly with wireline access and could include substantial levels of handoff (wireless links at both ends of a phone call). While handoff of cellular phone calls is not very common today, cordless-to-cordless telephone calls happen fairly frequently. Handoff produces two negative results: it doubles the delay due to the wireless system, and doubles the amount of speech distortion due to speech coding. Thus, if PCS handsets include private wireless access capabilities used in a residential and business cordless fashion, handoff of their digital speech coders could become common. While conversation is possible with round-trip delay approaching 400 ms, that level of delay will not be comfortable for many people. International Telecommunication Union — Telecommunication Standardization Sector (ITU-T) Recommendation G.114 suggests that one-way delay in national circuits be limited to 50 ms [14]. It also found that 10 percent or more of speakers may have difficulty when total delay reaches 400 ms. Prolonged conversations with voice quality that is only fair rather than good is also likely to be perceived negatively by the user. Providing speech quality comparable to wireline access, at least within a very limited radio range, has been an important strategy of the more successful cordless telephone equipment suppliers.

One approach to improving the voice quality of IS-136 technology is to improve the performance of voice coders at 8 kb/s. The IS-641 ACELP speech coding standard provides significant improvements over the initial IS-641 VSELP speech coder used with TDMA, but significant transmission delay remains, and tandem IS-641 coders do result in degraded performance.

A second approach to improving the voice quality of IS-136 technology is to allocate multiple time slots to individual users in order to support voice coders at 16 kb/s (two out of three time slots for one frequency channel) or 24 kb/s (all three time slots for one channel). Unfortunately, this has a substantial impact on capacity, impacts the ability of an economical terminal to perform the required signal strength measurements for mobile-assisted handoff, and requires terminals to include a duplexer.

A third approach to improving the voice quality of IS-136 technology is to introduce 8-phase shift keying (PSK) or 16-quadrature amplitude modulation (QAM) with efficient channel coding and diversity techniques to provide sufficient robustness in the presence of transmission impairments. This approach could permit the introduction of a higher-rate voice coder while maintaining the existing three traffic channels per carrier of IS-136. The 20 ms frame structure could be maintained while reducing transmission delay by removing interburst interleaving [15, 16].

**Speech Coding**

Two higher-rate speech coders were considered. ITU Recommendation G.728 and the GSM enhanced full rate (EFR) coder both give improved speech quality over IS-641. In this section both alternatives are considered and compared with IS-641 on the basis of complexity, voice quality, and delay.

The G.728 low-delay-CELP voice coder at 16 kb/s provides the possibility for an advanced mode of operation with toll voice quality and relatively low delay. The LD-CELP voice coder requires 30–45 MIPS processing power compared to about 20 MIPS for the IS-641 voice coder. Thus, digital signal processing (DSP) complexity with LD-CELP voice coding would be a near-term issue for an advanced voice mode of operation, but since the gap in complexity is modest and the performance of DSPs is growing rapidly, this issue would not be an obstacle very long. The GSM EFR speech coder at 12.2 kb/s was designed as a replacement for the original GSM speech coder in PCS-type systems. Its complexity is just slightly higher than the IS-641 speech coder due to a larger excitation search.

Wireless phones tend to be designed so that the microphone is further from the talker’s mouth than in fixed phones. They also tend to be used in higher ambient noise backgrounds than conventional wired phones used in the home or office. As a result, noisy input speech is more of a problem for wireless phones, and the performance of a digital speech coder in these conditions is significant in its acceptance in the marketplace. In addition to speech with music in the background, music on hold is an important quality characteristic for these coders.

If wireless phones become as popular as "cordless" phones for indoor use, there will be a greater and greater likelihood of encountering wireless-to-wireless phone calls. Handoff has two effects, both of them bad. Because the speech is encoded twice, the amount of delay the phone call encounters is doubled; and the amount of quantization degradation is also increased.
Table 1. Mean opinion score results for speech coded with background noise.

<table>
<thead>
<tr>
<th>Condition</th>
<th>Original</th>
<th>IS-641</th>
<th>G.728</th>
<th>GSM-EFR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Clean speech</td>
<td>4.34</td>
<td>4.09</td>
<td>4.23</td>
<td>4.26</td>
</tr>
<tr>
<td>Clean speech *2</td>
<td>3.62</td>
<td>3.99</td>
<td>4.13</td>
<td></td>
</tr>
<tr>
<td>15 dB babble</td>
<td>3.75</td>
<td>3.49</td>
<td>3.81</td>
<td>3.70</td>
</tr>
<tr>
<td>15 dB babble *2</td>
<td>3.08</td>
<td>3.49</td>
<td>3.79</td>
<td>3.47</td>
</tr>
<tr>
<td>20 dB car noise</td>
<td>3.72</td>
<td>3.61</td>
<td>3.64</td>
<td>3.76</td>
</tr>
<tr>
<td>20 dB car noise *2</td>
<td>3.11</td>
<td>3.58</td>
<td>3.48</td>
<td></td>
</tr>
<tr>
<td>15 dB office noise</td>
<td>3.70</td>
<td>3.40</td>
<td>3.61</td>
<td>3.58</td>
</tr>
<tr>
<td>15 dB office noise *2</td>
<td>2.75</td>
<td>3.55</td>
<td>3.31</td>
<td></td>
</tr>
<tr>
<td>15 dB music</td>
<td>3.89</td>
<td>3.82</td>
<td>3.98</td>
<td>3.99</td>
</tr>
<tr>
<td>15 dB music *2</td>
<td>3.16</td>
<td>3.92</td>
<td>3.85</td>
<td></td>
</tr>
</tbody>
</table>

Table 1 is a comparison of subjective test results of G.728, GSM-EFR, and IS-641 for clean and noisy speech with tandems. These results are from an Absolute Category Rating test run in April 1997 at AT&T Laboratories in Holmdel, New Jersey. (The conditions marked *2 indicate two encodings with the speech coders.) The results show an advantage for G.728 and GSM-EFR when compared to IS-641 for every condition. In these tests, the statistically significant difference was 0.16. The largest differences appear in these results for tandem conditions, while the differences for single encoding conditions are smaller. A comparison of the two higher-rate coders indicates that they are very similar in performance with no statistically significant differences.

Table 2. Delay in milliseconds for systems based on three different speech coders.

<table>
<thead>
<tr>
<th>Delay cause</th>
<th>IS-641</th>
<th>GSM-EFR</th>
<th>G.728</th>
</tr>
</thead>
<tbody>
<tr>
<td>Look-ahead</td>
<td>5</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Frame size</td>
<td>20</td>
<td>20</td>
<td>0.625</td>
</tr>
<tr>
<td>Processing</td>
<td>16</td>
<td>16</td>
<td>0.5</td>
</tr>
<tr>
<td>Bitstream buffer</td>
<td>0</td>
<td>19.375</td>
<td></td>
</tr>
<tr>
<td>Transmission</td>
<td>26.6</td>
<td>6.6</td>
<td>6.6</td>
</tr>
<tr>
<td>Delay</td>
<td>67.6</td>
<td>42.6</td>
<td>27.1</td>
</tr>
</tbody>
</table>

Table 2 is a comparison of the delays of the three coders. If we assume that a signal processor is chosen that can execute both the encoder and decoder combined in 80 percent of real time, we can make the assumptions that are made here for processing delay. Look-ahead is an extra number of samples used by IS-641 to form the linear predictive coding (LPC) analysis. This means having to buffer in an additional 5 ms of speech that will not actually be coded until the next frame. GSM-EFR and IS-641 both have 20 ms frame sizes, while G.728 has a five-sample (or 0.625 ms) frame size. Once the necessary speech is buffered, processing can begin. In the case of IS-641 and GSM-EFR, we will assume this takes 16 ms. For G.728, the first frame is processed in only 0.5 ms. However, the coder must then wait for the next frame to arrive until the complete 20 ms of speech is coded. Consequently, while IS-641 and GSM-EFR have 25 and 20 ms sample buffers that must be filled, G.728 must fill a 20 ms bitstream buffer. The last bits will fill that buffer 19.375 ms after the first bits are put into it. We have also indicated transmission delay. In the case of IS-641, we have included the frame interleaving that is part of the standard, while for G.728 and GSM-EFR we have eliminated it. (Frame interleaving is effective for fast fading channels, but ineffective for slow fading channels, which would be the case for indoor and pedestrian environments.) On this basis we come up with the total delays shown in the last row. This shows that G.728 does have a distinct advantage. Actual systems would have higher delay due to additional delays inherent in the system architecture.

In summary, the advantages of IS-641 over G.728 are in complexity and bit rate. G.728 is a low-delay coder and can produce better clear channel quality than IS-641 for a variety of SNR conditions.

Table 3. The performance of GSM-EFR error protection.

<table>
<thead>
<tr>
<th>SNR</th>
<th>Block error rate on protected bits</th>
<th>Bit error rate on protected bits</th>
</tr>
</thead>
<tbody>
<tr>
<td>9 dB</td>
<td>10^-4</td>
<td>2 x 10^-3</td>
</tr>
<tr>
<td>12 dB</td>
<td>8 x 10^-2</td>
<td>8 x 10^-2</td>
</tr>
<tr>
<td>15 dB</td>
<td>2 x 10^-2</td>
<td>6 x 10^-2</td>
</tr>
<tr>
<td>18 dB</td>
<td>8 x 10^-3</td>
<td>2 x 10^-2</td>
</tr>
<tr>
<td>21 dB</td>
<td>5 x 10^-3</td>
<td>8 x 10^-3</td>
</tr>
</tbody>
</table>
of conditions. GSM-EFR has clear channel quality performance on par with G.728, has a small delay advantage compared to IS-641, and seems like a good candidate as an upgrade to IS-641 for strong channel conditions. Either channel interleaving for low-mobility (slow-fading) channels will significantly reduce overall system delay.

Modulation, Channel Coding, and Diversity Techniques

In order to support the higher data rate requirement of the 12 or 16 kbps speech coders and to meet the requirement of three users in a 30 kHz channel, it is necessary to change the modulation from π/4-differential quadrature PSK (DQPSK) to a higher-level modulation such as 8-PSK [17, 18]. By changing only those symbols which would carry user data from 8-QAM to 8-PSK to higher-level modulation, the higher-layer protocols can be left largely unchanged. This would permit a high degree of commonality between various modes.

The maximum possible payload rate is 58.5 kbps with 8-PSK using the IS-136 format. This assumes that the same number of symbols is devoted to signaling and other overhead such as channel estimation as in the existing standard. The 58.5 kbps data rate with 8-PSK is enough to support up to 16 kbps for three voice users. Data users can be supported at 40–50 kbps with 8-PSK and multistage operation, and at 50–60 kbps with 16-QAM and multistage operation. The remaining data rate could be used for channel coding to provide robustness and for the additional channel estimation needed for data transmission in fast fading channel environments.

To provide protection against interference, fading, noise, channel coding and/or some form of diversity is necessary. A major benefit of channel coding is its ability to provide diversity in a fast fading environment, thus reducing the need for any other form of explicit diversity. Unfortunately, diversity through channel coding is not feasible in slow fading environments without excessive delay for interleaving or resorting to techniques such as frequency hopping. Since our focus is low-delay high-quality communications, large delay is not feasible, and any interleaving has to be restricted to one slot. Frequency hopping is not a part of the IS-136 standard, and would not provide significant benefit for IS-136 because of the long frame structure, which prevents hopping over a large number of frequencies within a channel interleaving period, so it is not considered further. Without these options, the coding can only provide some robustness against interference and thermal noise. For speech coders such as IS-641 or GSM-EFR (and less so for G.728), the compressed bits from the encoder exhibit differing sensitivity to channels. Hence, some form of unequal error protection is needed. Through subjective listening tests or more quantitative signal-to-noise ratio (SNR)-based bit error impairment tests, it is possible to determine which bits require more protection. We now describe an unequal error protection scheme for the GSM-EFR coder.

Frames consisting of 244 speech bits (representing 20 ms of speech data) are encoded and mapped into individual IS-136 slots that are 130 8-PSK symbols in length. The coding system works as follows.

First, the 244 speech bits are separated into three separate classes: class IA, consisting of the 84 most important bits, class IB, consisting of the 20 next most important bits (five for each of the four subframes); and class II, consisting of the remaining 140 bits (see [19] for details about the types of bits in the bitstream).

The 84 class IA bits are augmented with an 8-bit cyclic redundancy check (CRC), and then both the class IA and class IB bits are encoded together by a 32-state rate 1/2 convolutional code. The CRC checksum is used by the receiver to determine if a decoded frame is “good” or not. If the class IA bits satisfy the 8-bit checksum, the entire frame is considered usable (regardless of the number of undetected errors that may or may not be present in the class IB and II bits). If not, the frame is discarded and the receiving speech codec uses error concealment to interpolate over the lost frame.

A trailing set of five zeros is included at the end of the class I bits before encoding in order to terminate the convolutional code trellis. In total, 117 bits pass through the rate 1/2 code: 84 class IA bits, 8 CRC check bits, 20 class IB bits, and 5 trellis termination bits. The output of the convolutional code is therefore 234 output bits arranged in 117 pairs. Each of these 117 bit pairs select a “cloud” from the 8-PSK constellations shown in Fig. 2 for one of the first 117 symbols in the 130-symbol slot. The actual symbols that are transmitted within each cloud for these 117 symbols are determined by 117 of the uncoded class II bits. The remaining 140 – 117 = 23 class II bits are gray-mapped and encoded into eight separate symbols, and the remaining five symbols in the slot are left unused or can be used for channel estimation and tracking.

Figure 3 shows a block diagram of the complete coder. The performance of this unequal error protection scheme is shown in Table 3.

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**Figure 2.** 8-PSK constellation mapping for GSM-EFR.

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**Figure 3.** The error protection schematic for GSM-EFR.
Clearly, the SNRs required for high-quality performance (frame error rates much less than 1 percent) are significantly higher than in a cellular system with efficient frequency reuse. Additional SNR margin can be obtained using spatial diversity techniques.

**Enhanced Speed IS-136 TDMA Data Services**

TDMA provides 9.6 kb/s circuit data and fax access based on the IS-130 and IS-135 standards. Enhanced data rates will be important in the future, particularly for Web browsing on the Internet or on corporate Intranets. Enhanced speed access will require additions to the existing standards. The first step is likely to be multislot operation which can support rates up to 28.8 kb/s. A second step is to introduce over-the-air packet access for TDMA in addition to circuit mode access. A further step will be to introduce 8-PSK and/or 16-QAM modes to support data rates up to about 57.6 kb/s. Even higher bit rates may be possible based on transmit and receive diversity. Wireless data access to the Internet is likely to often come from stationary but widespread users with laptop computers or personal digital assistants (PDAs). Under those conditions, simple preselection diversity, as proposed for high-quality pedestrian voice service, could significantly improve downlink performance since fading rates will be slow.

The introduction of multislot operation brings several challenges:

- Increased signal processing requirements in terminals
- The possible requirement for a duplexer to support simultaneous transmit and receive for terminals
- A requirement to continue to support mobile-assisted hand-off (MAHO)
- Flexibility in the network to process both single-slot and multislot access

Since data-only operation removes the need to perform speech coding, which is a major signal processing demand for terminals, multislot data operation should not increase signal processing requirements significantly. Duplexer requirements are a concern for 1.9 GHz PCS operation, because the upband and downband are 60 MHz wide each with only a 20 MHz band separating them, so low-loss duplexers are a problem. This may motivate consideration of media access control (MAC) protocols which avoid requiring a terminal to simultaneously receive and transmit by scheduling periods at the BS for a terminal to transmit. A full-rate TDMA channel is arranged with one time slot out of three used for the terminal receiver every 20 ms and one time slot used for the transmitter that overlaps the two unused receive time slots. To avoid a duplexer requirement for multislot operation when the data transmission is primarily downlink, the MAC protocol could reserve two time slots (13.3 ms) every 100 ms for a terminal to acknowledge incoming data on the forward link. This would also provide time for terminals to make MAHO measurements. The cost would be a reduction in throughput by about 13.3 percent and an increase in latency due to sparse acknowledgments. Such a scheme also requires a protocol for varying the ratio of uplink and downlink bandwidth robustly in real time. One way to achieve robustness would be to make the BS the master with a fixed superframe of 100 ms and downlink transmissions beginning at the start of the superframe. A terminal would always be allocated one time slot at the end of the frame for transmission of acknowledgments, data, and control information, but could request more if required.

While circuit data evolution can take advantage of much of the network architecture, cell processing, and mobility systems developed for circuit voice service by adding interworking functions to wireline data protocols, a packet data capability for TDMA results in different requirements. Networks based on Generic Packet Radio Service (GPRS) are the most likely approach to support convergence of wireless data services for IS-136 TDMA and GSM networks. The Universal Wireless Communications Consortium (UWCC) Global TDMA Forum is considering packet data solutions for the TDMA air interface [20]. Channel assignment is a challenging problem for packet data over TDMA. A similar problem was addressed for enhanced TDMA (E-TDMA) in the past by Hughes for voice service with voice activity [21]. The solutions for packet data over TDMA are likely to be similar. Control complexity is increased for packet data compared to E-TDMA by a desire to support multislot operation as well as full-rate operation. For example, a solution is likely to include a control channel for packet operation on the downlink which assigns available channels to terminals as needed based on availability. Packet channel assignment would be based on pairing uplink and downlink channels as with circuit access in order to support acknowledgments and full-duplex operation. Since the BS is the master, uplink access during periods of downlink inactivity is likely to be based on a random access channel where a request is made for a channel assignment for uplink data. Very short uplink data transmissions may be accommodated directly on the random access channel. As for multislot circuit data operation, packet data access can include protocol provisions at the BS to schedule time slots for terminals to perform MAHO and acknowledgments without requiring duplexers at the terminal.

Multislot circuit data and packet data can be further enhanced for higher-speed operation by introducing higher-level modulation such as 8-PSK and 16-QAM to achieve data rates of 40 to 60 kb/s. This is based on assuming channel coding rates of about 5/6, which is the coding rate used for IS-130 (9.6 kb/s circuit data on IS-136). Figure 4 shows the throughput performance of the IS-130 radio link protocol (RLP). The RLP is assumed to have sufficiently large window size that the probability of window closure is small. The same rate 5/6 channel code proposed in IS-130 is used for QPSK, 8-PSK, and 16-QAM, resulting in efficiencies of 1.67 b/symbol, 2.5 b/symbol and 3.33 b/symbol, respectively. Assuming three data users in 30 kHz, the maximum data rates are 9.6 kb/s, 14.4 kb/s, and 19.2 kb/s. Ideal knowledge of the channel, no diversity, and perfect coherent demodulation is assumed in this study. As can be seen, it is advantageous to use 16-QAM over a wide range of SNRs. Below 18 dB, the performance of 16-QAM as well as 8-PSK rapidly deteriorates, and at 10 dB 4-PSK offers...
three times more throughput than 8-PSK or 16-QAM. In the presence of imperfect channel estimation, it is expected that 16-QAM and 8-PSK will be worse than 4-PSK at even higher SNRs. Furthermore, 8-PSK and 16-QAM will be progressively less robust to noise, interference, and delay spread compared to QPSK. Thus, mode adaptation is likely to be required which falls back to π/4-QPSK under difficult channel conditions. In order to achieve maximum benefit from the higher-rate modes, mode adaptation should be agile with little delay and minimal hysteresis based on a quality measure.

A large body of work exists on techniques for using 8-PSK, 16-QAM, and similar modulations over narrowband (e.g., 30 kHz) radio frequency (RF) channels. Differential modulation such as 8-DPSK, 16-DPSK, or 16-star-QAM have been proposed. In these schemes, the channel and rapid recovery from loss of carrier recovery during a deep fade [22–24]. Coherent detection has sometimes been proposed based on interpolation of pilot symbols or sequences [25, 26]. The combination of rapid fading due to mobile operation at 1.9 GHz resulting in fading at rates close to 200 Hz, and delay spread that can span under severe conditions up to about one symbol (41 μs) [27], is a difficult challenge for the use of higher level modulation. Simple numerical simulations have shown that with delay spread that spans 1/8–1/4 of a symbol [28]. Equalization is required for more severe delay spread, but it depends on coherent detection and accurate channel estimation. While a flat fading channel can be tracked using individual pilot symbols at a sampling rate beyond the Nyquist rate (500–600 Hz is adequate), estimating the rapidly fading dispersive TDMA channel using pilots or known symbols requires at least a few sequential symbols at each channel sampling point. Koipillai and Chennakeshu proposed a scheme based on block-coded modulation and equalization with known sequences sent at about a 300 Hz rate to support channel estimation for fading rates up to perhaps 100 Hz for cellular operation [29]. However, as the fading rate is increased to 500–600 Hz for 1.9 GHz operation, the overhead required for channel estimation becomes burdensome. An alternative is to track the channel after channel acquisition based on a sync word which undersamples the channel. Decision-directed algorithms (DDAs) are generally sensitive to bit error propagation, and non-decision algorithms such as constant-modulus algorithms (CMAs) generally track the channel more slowly. These problems may be mitigated by combining equalization and channel tracking with channel decoding and interleaving as proposed by Mehlan and Meyer [30].

The goal of delivering 40–60 kb/s over a 30 kHz TDMA channel will require 2.5–3.5 b/s/Hz performance. Techniques such as combined equalization and decoding may improve performance under difficult channel conditions, but adaptation to QPSK with slower transmission rates is likely to be essential to provide desired coverage levels. Simple preselection diversity for terminals is desirable for stationary operation of laptop computers and PDAs, particularly since Web browsing applications will stress downlink performance. Providing multitask and circuit and packet data access may be possible without introducing terminal multiplexer requirements with modest penalties in data throughput.

**Smart Antenna Technology for 15–156**

Smart antennas and adaptive antenna arrays have recently received increasing interest to improve the performance of cellular radio systems. Smart antennas include a large number of techniques that attempt to enhance a desired signal and suppress interfering signals. While adaptive array antennas and steerable beam antennas have been used for military applications for decades, only in the last few years have these techniques begun to be applied to commercial systems.

Research on adaptive antenna arrays for cellular systems dates from the early to mid-1980s [31, 32], but R&D on smart and adaptive antennas for cellular systems has intensified only in the last few years. In 1995, Nortel introduced smart antenna technology for PCS-1900 systems [33]. Other companies such as Metawave have introduced similar technology [34], and the European Advanced Communications Technologies and Services (ACTS) TSUNAMI project is considering adaptive antennas for third-generation wireless systems [35]. A number of R&D efforts have also considered adaptive antennas, and the TDMA standard is incorporating a few [36].

Smart or adaptive antennas are receiving strong interest because of several critical factors. The cellular networks at 850 MHz in the United States are becoming increasingly crowded, and smart antennas offer the possibility of improving quality while increasing capacity by suppressing interference and supporting operation with lower cellular reuse factors. The new PCS bands at 1.9 GHz are subject to increased path loss relative to the cellular bands that approaches 10 dB. This means that the new PCS systems will be under significant stress to provide adequate coverage, and techniques to extend range will be very important. Since wireless access has become dominated by handsets with limited transmit power, the uplink is generally the limiting factor for range. Finally, the key technologies required for smart antennas, DSP, and efficient RF systems are rapidly advancing. DSP chips are becoming available with throughput on the order of 100 MIPS that require only hundreds of milliwatts of power, which means that smart antennas depending on complex algorithms are becoming commercially feasible. The cost, size, and capabilities of RF systems are advancing. For instance, wideband digitization and digital receiver techniques are becoming practical for BSs, and high-power multicarrier chip sets are now attractive alternatives to single-carrier amplifier and cavity-combiner approaches. Smart antennas offer the potential to lower the cost of cellular infrastructure. One estimate suggests that smart antennas can reduce the cost of infrastructure for a cellular or PCS system by up to 60 percent [43].

There are two basic classes of smart antennas being considered for cellular systems:

- **Switched beam antennas**
- **Adaptive antenna arrays**

A typical switched beam antenna has four 30° beams within a conventional 120° sector for a BS. This antenna consists of four collinear elements placed in a line in front of a reflector to individually form 120° patterns. The collinears are spaced at about one-half wavelength apart and are fed through a four-port Butler matrix. This arrangement provides four 30° beams, each with a peak gain about 6 dB higher than any collinear individually. By combining such an arrangement with an RF switching matrix and a scanning receiver to choose the beams with the best signals from a mobile, uplink sensitivity can be improved for a desired signal, and interference is suppressed that is outside the selected beams. A variation on this approach for cellular and PCS applications is to replace the Butler matrix with processing that allows the peak of a beam to be pointed at any angle within the 120° sector.

In the adaptive antenna array case, four conventional 120° sector antennas may be spaced uniformly with a total aperture of 10–20 wavelengths (5–10 ft at 1.9 GHz). A goal of this antenna configuration is to achieve minimum correlation in fading between the four elements to maximize diversity gain. The probability of simultaneous fades is then low. The signals
from the four antenna elements are individually processed and combined using DSF. The four signals are combined using weights that maximize the desired signal and suppress any interfering signals. There are many possible algorithms for generating the combining weights, and also many different antenna element configurations that could be considered. Other possible antenna configurations include using two dual-polarized antennas with 120° beamwidths.

The switched beam antenna approach has the potential advantage of a small aperture compared to the two-branch spatial diversity arrangement generally used for cellular or PCS BS receivers, which uses two antennas spaced by 5–10 ft. However, a single switched beam antenna with four beams per sector will only support about the same link budget as a comparable full sector spatial diversity arrangement with two antennas. The two-branch spatial diversity arrangement typically provides 4–6 dB improvement in signal quality in a fading environment [44], and the switched beam antenna provides up to 6 dB peak gain over a conventional antenna within a beam and the possibility of angle diversity by combining signals from different beams. In many macrocellular environments, such as rural and suburban, the angular spread of the received signals at a BS is usually much less than 30°, so the switched beam antenna only provides one strong signal much of the time, and is subject to Rayleigh fading. In these environments, the switched beam antenna provides little angle diversity gain [45]. Significant range extension of 4–6 dB compared to two-branch spatial diversity can probably be achieved by combining switched beam antennas with other diversity arrangements. For example, two switched beam antennas could be spatially separated by 5–10 ft at 1.9 GHz, or possibly two switched beam antennas could be collocated within a small aperture using polarization diversity.

With regard to interference, switched beam antennas can provide some attenuation of any interfering signals that are not arriving at the BS within the beam or beams selected for the desired signal. However, for the single switched beam antenna, the receiver requires a carrier-to-interference ratio (C/I) about 6 dB higher than that for the two-branch spatial diversity receiver for interferers located within a selected beam for environments with little angular spread. For example, for IS-136 TDMA a C/I of about 17 dB is required for acceptable quality in a fading environment without diversity reception, but a C/I as low as 11 dB provides acceptable quality with diversity reception with two equal-strength independently fading received signals. Thus, the single switched beam antenna will provide some rejection of interference outside of selected beams, but is more vulnerable to interference within selected beams than conventional two-branch spatial diversity. Overall, the single switched beam antenna probably provides about equivalent performance against interference as conventional two-branch spatial diversity.

The adaptive antenna array approach offers the potential to strongly suppress interferers and to provide significant range extension. At the .01 BER point, ideal two-branch diversity has about 6 dB advantage over a one-branch system, and four-branch diversity has about 6 dB additional advantage over the two-branch diversity case. Field test results based on a system described in [51] by Cupo et al. shows about 4 and 8 dB gains in performance for two- and four-branch diversity approaches compared to 6 and 12 dB gains for ideal results, primarily due to correlation in fading between antennas. Figure 3 shows field test results of the performance of an IS-136 signal against noise in a Rayleigh fading environment with the additional impairment of an interferer with average signal strength equal to the desired signal. Even with C/I = 0 dB, the four-branch adaptive array is able to obtain a BER of about 0.01.

Thus, a four-branch adaptive array offers the potential to provide up to 6 dB range extension over a conventional two-branch antenna system. If the signal attenuation is assumed to increase on the average at a rate proportional to the fourth power in a cellular or PCS environment, a 6 dB range extension will increase the range by the square root of two and decrease the required number of BSs by a factor of two to satisfy coverage constraints. In a realistic environment with significant correlation in fading between the signals received on a four-element antenna array, the gain in range extension may fall to about 4 dB, which means that an area can be covered with about two-thirds the number of BSs using a four-element adaptive antenna array compared to a conventional two-branch spatial antenna diversity system.

With IS-136 TDMA, each burst is 6.7 ms long. At 1.9 GHz, a fading rate of 184 Hz corresponds to 60 mph. This means that several fades can occur within a single burst, so the adaptive antenna array must track the channel over the burst for the desired signal as well as for any interfering signals. The Direct Matrix Inversion (DMI) approach [36] was shown to provide good performance in tracking rapidly fading IS-54/IS-136 TDMA signals in the presence of interference using a minimum mean square error (MMSE) solution. The signal is tracked over a burst one symbol at a time by using a decision directed mode of operation to generate a reference signal for the entire burst.

Additional problems the adaptive antenna array must consider include the effects of a dispersive channel and the effects of an asynchronous interferer. The effects of a dispersive channel can be addressed by using either a linear or decision-feedback equalizer in combination with a spatial array [46] and DMI, or MMSE in combination with DMI [47]. The effects of an asynchronous interferer (an interferer that turns on after the synchronization sequence) can be minimized by using an algorithm with rapid tracking capabilities.

IS-136 TDMA is based on FDD operation, so the uplink and downlink signals generally fade independently, and determining weights for a downlink transmit adaptive array becomes problematic. One solution is to place multiple receivers at the handset, but that may not be economical. Another proposal is to use angle-of-arrival algorithms for an uplink and form downlink beams at that estimated angle [48]. These techniques have been also extended to include null steering of the BS transmit array based on the received signals to reduce interference [49]. A disadvantage of this approach is the requirement for transmit paths that are well phase calibrated to the different elements of the adaptive array, which is a concern for conventional systems with antenna elements on a tower connected to RF circuitry at the base of the tower by individual coaxial lines. Also, transmit null beam forming for FDD systems is likely to be sensitive to angular spread of the signal. Another proposal for adding smart antenna tech-
unities to downlinks uses switched beam approaches where a beam for the downlink is assigned based on a beam that is selected for the uplink.

For IS-136 TDMA, downlink smart antenna approaches are further complicated by the requirement that downlink signals be locally continuous in time, which precludes switching or adapting beams in a step function between time slots. One approach would be to modify the standard to support a discontinuous downlink so that beam-switching could occur between time slots. However, many terminals depend on using both the synchronization word for the desired burst and the synchronization word from the following burst. A good approach to this problem is to transmit on all downlink beams using independent power control for each beam that satisfies the power requirements for any and all terminals in each beam. Changes to power control levels would need to be applied with smooth trajectories that do not violate the requirement for local signal continuity. The IS-136 standard includes a provision for terminals reporting over the uplink 1 s averages of received signal strength and BER of the downlink as seen by the terminal. Analysis of this approach shows that frequency reuse can be reduced to four from seven while improving quality.

While switched beam antenna technology appears to be the most feasible for IS-136 downlinks, adaptive antenna array technology appears to be the most attractive for uplinks. A smart antenna for IS-136 systems with significantly improved uplink and downlink performance over conventional systems may be best attained by a hybrid approach that combines switched beam antennas with power control for the downlink, and conventional antennas and adaptive array processing for the uplink. An antenna configuration that satisfies this requirement for a conventional 120° sector would consist of a single switched beam antenna with four 30° beams for transmission and two dual-polarized 120° antennas spatially separated from the switched beam antenna for reception.

Another alternative for downlink diversity that has emerged in recent years is space-time coding [50, 51]. Here, the data is encoded by a channel coder, and the encoded data is split into two parallel streams and sent over two transmit chains. The purpose of the channel code is to select the sequence of code symbols to be transmitted simultaneously from each antenna in order to maximize immunity to fading and noise. At the receiver, maximum likelihood decoding is used to provide diversity gains against fading and coding gain against noise.

A simple example of a space-time code is to transmit the same signal from both transmit chains but with a delay of one symbol between them [52, 53]. We can view the space-time code as a simple repetition code with a one-symbol transmission delay between code symbols. However, such a code provides only diversity gain. It has been shown [50, 51] that better codes can be found which provide additional 4-5 dB coding gain.

DCA schemes have been implemented for indoor, low-tier or pedestrian technologies such as DECT and PHS [56, 57]. The use of per-carrier amplifiers and cavity combiners at BSs, which is still common in cellular systems, has been a major impediment to the introduction of DCA for cellular systems. Other impediments include difficulty in implementing DCA with very large systems, performance under heavy load conditions, and challenges of evolving from FCA to DCA. However, several factors suggest that DCA will become a core capability for TDMA systems:

- The emergence of hierarchical cellular systems
- The development and deployment of automatically tuned cavity combiners and multicarrier amplifiers
- The experience gained from early adapters of DCA techniques

With IS-136 TDMA, since the downlink is continuous and frame structures are typically not synchronized between BSs, entire carriers should be borrowed or dynamically allocated to a particular BS. Along with other factors, this provides motivation for packing users onto a minimum number of required carriers in order to provide a maximum pool of free carriers and hence channels in a given local area that can be dynamically assigned.

TDMA uses many channels, each only 30 kHz wide. This makes DCA particularly well suited to DCA for microcells, PBSS, and wireless office systems within a macrocellular system that uses either FCA or DCA. With channels only 30 kHz wide, microcells and indoor systems that are uncoordinated or loosely coordinated with a macrocellular system can assign channels in small increments. This is particularly useful for the coexistence of a large number of overlapping or nearby systems that are uncoordinated except for frequency assignment arrangements. If microcells and indoor systems make measurements to avoid channels used on nearby macrocells and to coordinate frequency use among themselves, this will motivate some rules on DCA to minimize channel assignment churn and harmful interference. For example, macrocellular DCA should include some rules or inertia in reassigning channels to avoid churn in microcellular and indoor system frequency assignments. Microcells typically transmit with much higher power than microcells, so microcells can readily measure the signals of nearby macrocells for ACA, but macrocells cannot easily measure the signals of nearby microcells, so macrocells must depend on rules or databases instead of measurements to avoid interfering with microcells when making new channel assignments. For example, one proposal excludes a number of channels randomly selected or based on a predetermined list at each macrocellular BS from use in DCA [58]. As long as the exclusion list is not too large, there is very little impact on the efficiency of DCA for the macrocellular system, but it permits wireless office systems and other small cell systems to find channels that are unused over the long term.

Two basic types of DCA schemes have been studied for cellular applications:

- Measurement-based schemes that make real-time measurements at the mobile and/or BS to find “unused” channels
- Network-based schemes that use a set of rules to control interference by maintaining minimum reuse distances or a learning process for finding good channels

With certain assumptions, measurement-based schemes can provide very high performance, particularly if channel selection is based on a combination of both mobile and BS measurements. For example, in one proposal each BS broadcasts a candidate list of low interference channels as seen by its receivers [59]. A mobile scans the BS's candidate list for the lowest interference channel as seen by its receiver, so the resulting channel selection process provides a low interference channel for both the uplink and downlink. A common assumption for measurement-based DCA is that all terminals are rela-

Automatic and Dynamic Channel Assignment for IS-136

Today's TDMA systems are almost entirely based on conventional fixed channel assignment (FCA). The exception to this are emerging applications of wireless office systems, PBSSs, and microcellular systems. On the other hand, a large body of work exists which has examined the possible use of automatic channel assignment (ACA) and dynamic channel assignment (DCA) for Advanced Mobile Phone System (AMPS), TDMA, and other cellular systems, and DCA has been considered for mobile systems from the early days of research on cellular telephony [54, 55]. The motivations for DCA include spectrum efficiency, dealing with hot spots, and reducing manual frequency planning.

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tively stationary. For a given set of channel assignments using measurement-based DCA, the optimality of those channels degrades very quickly for mobiles where shadow fading can change in a few seconds. Existing circuit-based cellular systems, including IS-136, require handoffs in order to make a new channel assignment, but handoffs are typically limited to rates of a few per minute per mobile, so measurement-based DCA cannot keep up with changes in shadow fading for IS-136 which can change in a few seconds. Thus, network-based DCA schemes are appropriate for IS-136. Measurement-based DCA schemes may be good candidates for low-mobility systems, fixed wireless access systems, and next-generation mobile packet systems that reassign channels on a per-packet basis.

Channel borrowing is one of the most basic and lowest-impact network-based DCA schemes [60]. It assumes that a conventional fixed assignment exists, but that channels can be "borrowed" from nearby cells when such borrowing would not violate minimum reuse distances for any active calls. Channel borrowing can reduce efficiency under heavy load conditions by preventing future calls from being completed on borrowed channels in their "home" cells, but channel borrowing can improve performance under light to moderate loading conditions [61]. A number of variations of channel borrowing exist with trade-offs in performance. Basic characteristics of channel borrowing are that a frequency reuse plan is required as for conventional fixed reuse, and it allows for a modest detriment operating transition from conventional fixed reuse. Channel borrowing could be combined with exclusion sets for macrocellular DCA to support underlaid microwells and wireless office systems obtaining channels based on their measurements.

Centralized and distributed DCA schemes are based on either maintaining a central pool of channels that are assigned as needed and then returned to the pool while obeying minimum reuse rules, or assigning channels from the set of all possible channels based on knowledge of channel usage only for neighboring BSs within the minimum reuse distance. Local packing is a simple example of a distributed DCA scheme [62]. Centralized and distributed DCA schemes could also be combined with exclusion sets for macrocellular BSs to support hierarchical cellular operation. Adaptable DCA schemes consider the recent history of traffic, not just the present state of channel usage, to determine channel assignments, which builds some inertia into the channel assignment process.

DCA for IS-136 evolution is likely to be based on either channel borrowing, adaptive DCA, or distributed DCA with some rules to support hierarchical cellular operation. DCA for macrocells is likely to be based on minimum reuse rules, and microcells, wireless office systems, and PBSSs are likely to combine ACA based on measurements to find channels not used by nearby macrocells and DCA within such systems.

**Microcellular Evolution for IS-136**

Microcells today share much of their architecture with macrocells. In the future, TDMA microcells will be more highly optimized for small cell operation based on several trends, including:

- Centralization of functionality
- Miniaturization
- Power reduction
- Flexibility of interconnections
- Line powering for picocells
- Automatic channel assignment

The optimization of microcells for small cell operation will make TDMA more cost effective and flexible for deployment in dense pedestrian areas and private indoor environments.

For example, high-speed digital subscriber line (HDSL) is emerging as a low-cost technology for provisioning T1 service over ordinary unloaded copper loops for distances up to about 12,000 ft, and may be an example of a macrocell in the future. An HDSL transceiver pair now costs less than US$1000 and is likely to fall to half that over the next three to five years. A single TDMA BS transceiver can be served over a 64 kb/s link, so an integrated services digital network (ISDN) access line with two 64 kb/s circuits can serve two BS transceivers in a microcell arrangement.

Advances in high-speed analog-to-digital (A/D), digital-to-

analog (D/A), and DSP technology, and availability of high-speed fiber networks may permit microcells that contain only RF processing, A/D, D/A, and fiber interconnect circuitry and support the digitization of an entire RF band of operation. Possible advantages include microcells that are independent of modulation and standards evolution. A significant challenge to this approach is the required bandwidth for microcell fixed interconnection, which is two to three orders of magnitude higher than the aggregate data rate supported to end users by a microcell.

Another approach that is sometimes considered is remoting microcells over fiber/coax networks using analog transmission techniques. The major challenges with this approach are dynamic range for the fiber/coax network and required bandwidth on the upstream.

### Future High-Speed Packet Data Wireless Access Using EDGE

International Mobile Telecommunications in the year 2000 (IMT-2000) proposes to provide a broad range of services. Web browsing and information service access, which have caused the recent explosion in Internet usage, are highly asymmetrical in transmission requirements. Only the transmission path to the subscriber need be high-speed for many applications. Many other services provided over the Internet can also be provided with low to moderate bit rate uplinks. However, large file transfer is an example of an application which benefits from symmetrical high-speed transmission. This section describes an evolutionary path for high-speed packet data for TDMA systems.

The UWCC has adopted Enhanced Data rates for GSM Evolution (EDGE) based on enhancing GSM packet data technology with adaptive modulation to support packet data access at peak rates up to about 384 kb/s [63]. This system will adapt between Gaussian minimum shift keying (GMSK) and 8-PSK modulation with up to eight different channel coding rates, as shown in Table 4, to support packet data communications at high speeds on low-interference/noise channels and at lower speeds on channels with heavy interference/noise. EDGE can be deployed with the conventional 4/12 reuse used for GSM.
However, the use of 1/3 reuse is also proposed to permit initial deployment with only 2 x 1 MHz of spectrum. A combination of adaptive modulation/coding, partial loading, and efficient automatic repeat request (ARQ) permits operation with very low delay factors. Incremental redundancy or hybrid ARQ has been proposed to reduce the sensitivity of adaptive modulation to errors in estimating channel conditions and improve throughput [64]. One challenge for 1/3 reuse is the operation of control channels which require robust operation. This can be accomplished by synchronizing the frame structures of BS transceivers throughout a network and using time reuse to achieve adequate protection for control channels. GPRS networking is proposed to support EDGE-based wireless packet data access.

Conclusions

This document considers evolutionary changes to IS-136 TDMA to enable it to provide a variety of third-generation wireless services. These evolutionary changes include options that would:

- Provide high-quality voice service for indoor and pedestrian systems such as cellular office systems and personal base stations
- Support enhanced bit rate packet wireless data access to the Internet as well as circuit data access
- Provide smart antenna technology to improve coverage, quality, and capacity
- Automatically assign frequencies for operation and provide for dynamic channel reconfiguration
- Support microcellular arrangements to provide low-cost high-capacity service in dense areas
- Support future high-speed packet data access modes through a wideband system that complements IS-136 TDMA

The introduction of a complementary high-speed packet data service will require very high spectrum efficiencies to permit the economical refarming of spectrum from existing narrowband services to high-speed operation. Smart antenna technology to support IS-136 appears to be a viable option to improve coverage by 4 to 5 dB, frequency reuse by about a factor of two, and quality of IS-136 systems while maintaining compatibility with the existing standard. Higher-quality and lower-delay speech for indoor and pedestrian environments and higher-rate data transmission are feasible by introducing enhanced modes to IS-136 within the existing 30 kHz channel with minimal new requirements on terminals and base station equipment.

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References

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