VOICE CAPACITY ENHANCEMENTS FOR GSM EVOLUTION TO UMTS

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White Paper Developed for

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Introduction

GSM has proven itself as the most effective cellular technology worldwide, becoming the de facto standard with 438 GSM networks deployed in 157 countries. GSM provides global roaming, huge economies of scale, a clear migration path to next-generation systems, and capabilities today that match or exceed competing technologies. Constant innovation of GSM technology has continually increased GSM voice capacity, and the evolution continues with both EDGE and UMTS/WCDMA providing significant new features and services. This paper describes some of the groundbreaking methods and enhancements used by the latest GSM that maximize capacity to greater than sixteen times AMPS today, and thirty-two times AMPS in the near future. It compares results against TDMA, CDMA2000, and UMTS systems, and describes enhancements available with EDGE and UMTS.

Overview

The International Telecommunications Union approved EDGE (via UWC-136), UMTS/WCDMA, DECT, TD/SCDMA, and CDMA2000 as official third-generation cellular standards. Yet, what the industry needs to better understand is that the sophistication and efficiency of GSM technology makes it perform on par with third-generation systems.

In fact, GSM has become one of the most spectrally efficient (highest-capacity) technologies for wireless access[1]. While CDMA technology is often used for wireless access, such as in UMTS, IEEE 802.11 wireless LANs and CDMA2000, GSM technology should be measured alongside it for providing high capacity. The key technological concept common to most high-capacity approaches is that of *spread spectrum*. In spread spectrum each cell site spreads its radio signals across the entire spectrum bandwidth, and thereby shares this scarce resource with other nearby base stations. In this manner, the spectrum is re-used in every cell, allowing only the interference seen by a given cell to dictate the capacity of that cell.

CDMA, in fact, is a spread-spectrum radio technology that uses different "codes" for spreading the signals in a given cell site across a certain amount of radio spectrum. In addition to the spectrum being re-used in every cell site for capacity purposes, an added advantage of spreading is that the radio signal is more immune to signal fading[2] effects, since fading is specific to certain frequencies, and the frequencies do not all fade together.

GSM with the use of frequency-hopping techniques, however, is also a spread-spectrum system, and has the same advantages normally attributed to CDMA. Additionally, in the case of GSM the spread spectrum is not limited to 1.25 MHz but can use all the available bandwidth. Consequently, GSM combined with frequency hopping, advanced radio planning, and new voice compression methods boost GSM capacity to equivalent to, or better than, CDMA2000.

Some of the specific GSM innovations described in this paper include:

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[1] *Access* denotes the radio technology used between mobile terminals and base stations.

[2] *Fading* is when the random phases of the radio signal add constructively or destructively producing a signal amplitude variation as a function of spatial location and user velocity.

- **Adaptive Multi-rate speech transcoding (AMR).** This codec allows dynamic management of voice quality and error control, providing good voice quality even under adverse radio conditions.
- **Frequency hopping.** By rapidly changing the frequency of voice traffic channels, frequency re-use can be tightened to re-use the frequencies in every cell site, or in every sector (referred to as N=1 or 1/1 re-use).
- **Dynamic power control.** By constantly adjusting transmission power levels of mobile terminals, GSM reduces interference and increases capacity.
- **Discontinuous transmission.** GSM further increases capacity by reducing transmission when users are silent during their voice calls.
- **Future innovations.** GSM will increase capacity even further with Dynamic Frequency and Channel Allocation, Single Antenna Interference Cancellation (SAIC), the EDGE radio interface, and ultimately UMTS.

What matters is not what generation a technology is labeled, nor the type of technology used, but what capabilities and features the technology can deliver. Once capabilities are properly understood, one should consider which technology offers better costs and economies of scale. This is where GSM, which represents over 80% of the global cellular market, has an overwhelming advantage both due to higher volumes and lower inherent equipment costs, even when the innovations described in this paper are applied.

Voice Capacity Definition and History

There are two common ways in which voice capacity is measured. One is in *Erlangs*, which refers to carried traffic over a certain period under a particular grade of service (such as blocking rate[3]). The other is in *active voice paths*, referring to the total number of voice paths available at any one moment in time. These are sometimes confused, but there is a significant difference as they measure different items. For example, a sector with one hundred voice paths that needs to have a blocking rate of two percent (2%) can have only eighty-six voice paths active on average over time. Thus, over an hour, this sector would carry eighty-six Erlangs. In this paper we use *Erlangs* in combination with grade of service.

It is worth commenting on claims made about different technologies. CDMA technology has consistently been presented under quite optimistic assumptions. When first introduced, vendors promised over a twenty-fold increase in capacity over the analog (Advanced Mobile Phone Service or AMPS) network. Once deployed, actual gains were closer to five or six fold. Only later with the Enhanced Variable Rate Coder (EVRC) did an improvement factor of ten to twelve materialize, still only half of the original claim. In contrast, the GSM community has been more conservative with its estimates. The figures used in this paper are based on multiple sources and are defendable.

[3] *Blocking rate* refers to the percentage of call attempts that fail due to the system being at capacity.

GSM Spectral Efficiency Solutions

This section explains in technical detail the mechanisms used and planned for GSM that increase spectral efficiency. Some of these are already commonly used. Others will be available soon, and others are still under development. Of these, AMR and frequency hopping provide the greatest immediate benefit. The other methods work in harmony with these to help realize optimal gains. Beyond the immediate solutions, new technologies such as SAIC promise significant additional improvement.

This paper provides capacity-improvement estimates for each mechanism individually in the following sub-sections. While combining of multiple mechanisms provides greater improvement, the improvements are not always completely additive.

The evolution of GSM capacity-improvement methods is as follows.

Evolution Stage	**Capacity Improvement Method**
Earliest	Basic time-division multiplexing
Current base line	Frequency hopping Tighter frequency re-use patterns Power control Discontinuous transmission
Current advanced methods	Re-use Partitioning
Emerging and proven	Adaptive Multi-rate (full-rate and half-rate) toll-quality voice-coding Dynamic Frequency and Channel Allocation
Under development	Single Antenna Interference Cancellation 8-PSK voice for more robust half-rate operation

Table 1: GSM voice capacity evolution

Adaptive Multi-rate Codec

The Adaptive Multi-rate (AMR) codec is one of the most important innovations for GSM. Once widely deployed within a system, it is expected to more than double voice capacity. AMR will not be used only in GSM, but also in EDGE and UMTS networks. It is designed to work with both GSM full rate (one user per each of eight time slots in each radio channel) and GSM half rate (two users per time slot). AMR defines multiple voice encoding rates, each with a different level of error control. The AMR codec dynamically responds to radio conditions, using the most effective mode of operation at each moment of time. Compared to the Enhanced Full Rate (EFR) codec, AMR can operate under much worse radio conditions, such as with a heavily loaded network. AMR benefits include:

- Greater spectral efficiency, hence higher capacity, from tighter frequency re-use with frequency hopping.

- Better voice quality throughout the cell, especially at cell edges and deep inside buildings, and increased overall coverage.
- The potential of operating with toll-quality voice in half-rate mode, which reduces network costs.

The eight AMR full rate modes operate at the following rates (in Kbit/s): 12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15 and 4.75. The 12.2 Kbit/s rate is the same as the current GSM EFR codec, and the 7.4 rate is the same as the ANSI-136 TDMA codec. The gross bit rate of the channel (1 time slot) is 22.8 Kbit/s, which is divided into voice information and error control. As an example, 7.95 Kbit/s mode means that more than half of the bit rate (approximately 22.8 minus 7.95 minus some overhead) can be allocated to channel coding (forward error correction). By decreasing the AMR rate, resistance to errors increases further. Figure 1 illustrates this trade-off between voice coding and error control. Figure 2 shows the resulting affect from voice quality (Mean Opinion Score) versus the Carrier-to-Interference (C/I) ratio.

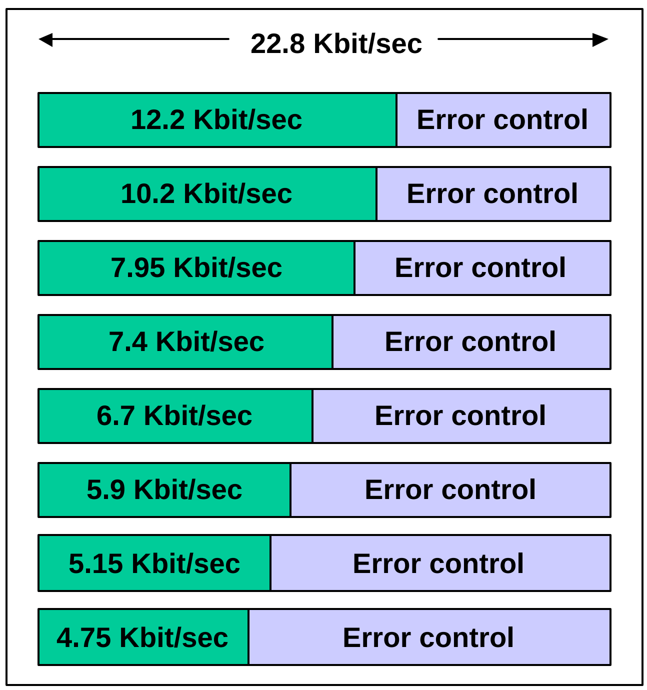

Figure 1: Ratio of voice encoding rates relative to error control with AMR full rate

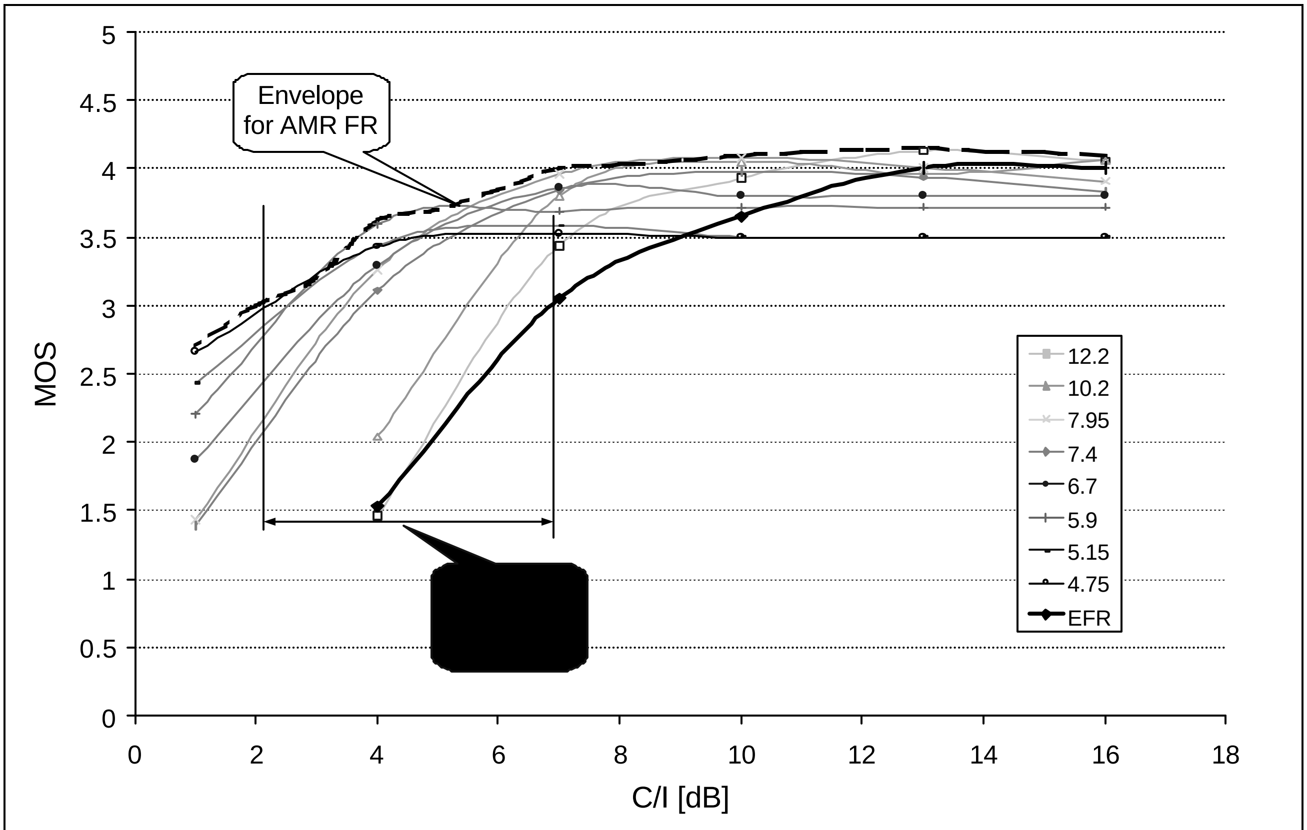

Figure 2: The effect of Adaptive Multi-rate vs. C/I [4]

This dynamic capability means AMR can compensate for the higher error rates arising from techniques such as tighter frequency re-use and higher fractional loading, which inherently force the mobile terminal to operate at lower C/Is.

The six AMR half rate modes operate at the following rates: 7.95, 7.4, 6.7, 5.9, 5.15, and 4.75 Kbit/s. Since the gross bit rate in a half-rate channel is only 11.4 Kbit/s, a much smaller number of bits is available for channel coding, thus requiring a better C/I. But with a better radio signal, AMR can enable half-rate operation, which translates to more users in the same number of radio channels. AMR half-rate mode is further enhanced in EDGE radio networks where more bits per time slots are available.

The benefits of AMR do not depend on all mobile phones implementing AMR. As the percentage of mobile phones with AMR increases in the network, the efficiency of the network increases. For instance, with 50% of mobile telephones using AMR, voice capacity can increase by 50%, whereas with 100% AMR use, voice capacity can increase by a full 150%. Carriers will begin selling mobile telephones with AMR in 2002.

[4] Reprinted with permission of the publisher John Wiley & Sons, Ltd. from "GSM, GPRS and EDGE Performance." Copyright 2002 by John Wiley & Sons, Ltd. This book is available at all bookstores, www.amazon.com, and at www.wiley.com, or call (732) 469-4400.

Frequency Hopping

Operators have used frequency hopping since the mid 1990s to mitigate interference. In combination with other techniques such as fractional frequency loading and AMR, frequency hopping enables tighter frequency re-use, down all the way to N=1, in which the same frequencies are re-used in every sector much the same as in CDMA. This significantly improves the capacity of GSM networks as it allows a greater number of radio channels to operate in each sector.

In frequency hopping, GSM changes the carrier frequency every TDMA frame (which consists of eight time slots over a 4.6 msec period), selecting from a pool of frequencies. The sequence of frequencies is pseudo-random. Since interference is typically frequency specific, only a small amount of information is lost, and the initial transmission can be recovered using standard coding techniques such as forward error correction and interleaving. Frequency hopping yields substantial link-level gains, ranging from 6 to 8 dB, and can double voice capacity.

There are two fundamental approaches to frequency hopping. One is called base-band and one is called synthesized frequency hopping. In the base-band approach, the number of transmitters at the cell site is equal to the number of hopping frequencies. Each of these transmitters operates at a fixed frequency. On a frame-by-frame basis, calls are switched to each particular transmitter.

Synthesized frequency hopping is more sophisticated in that the radios synthesize the transmission frequency of each TDMA frame. Unlike the base-band approach, a voice call remains associated with one transmitter. This approach allows a broader number of frequencies to be employed in the hopping pattern, thus increasing the benefits of frequency diversity and providing greater flexibility in managing the overall frequency plan. The ratio of transmitters that hop to the number of hopping frequencies is called the *fractional load* (also called *frequency load*), and represents the mean usage of a transmission frequency.

The effectiveness of frequency hopping increases with the total amount of available spectrum, as this provides greater interference diversity. Also, frequency hopping is not usually applied to the channels employing the control channel, and this portion of the spectrum is referred to as the non-hopping layer. The more spectrum that is available, the greater the percentage of it that can be assigned to frequency hopping. This portion is referred to as the *hopping layer*.

In the non-hopping layer, which contains the broadcast control channel (BCCH), operators deploy traditional re-use patterns such as 4/12[5]. In the hopping layer, operators can deploy tighter re-use patterns such as 3/9, 1/3 and 1/1. In the 1/1 re-use pattern, the entire pool of hopping frequencies is available in every single sector. In combination with other methods, such as discontinuous power transmission, dynamic power control, and AMR, the number of actual frequencies used (the fractional loading) can increase, thus increasing voice capacity. For instance, without AMR, fractional loading radios are limited to the range of 20% to 30%, but with AMR, carriers can achieve fractional loads of 40 to 60%.

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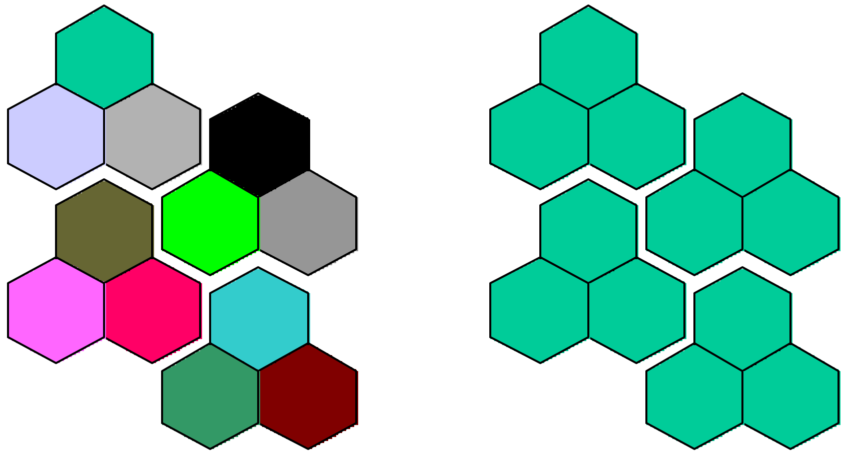
[5] *4/12 re-use* means that available radio channels are used across four cells, each with three sectors. Each sector has 1/12 of the total channels. The pattern is repeated every four cells.



4/12 Re-use. 4 cell sites each with three sectors. 12 groups of radio channels.

1/1 Re-use. Each sector uses the same pool of radio channels.

Figure 3: Tighter re-use enabled by frequency hopping

Re-use Partitioning

This method refers to having different layers in each cell sector with different levels of re-use. Originally, this concept did not depend on frequency hopping. But most implementations today take advantage of frequency hopping in the higher-capacity layers.

The non-hopping layer has broader coverage, and operators can use this layer to provide continuous coverage. This is sometimes referred to as the *overlay*. Operators can then use a separate layer that is usually based on frequency hopping to increase capacity. This layer does not necessarily have the same coverage as the non-hopping layer, and sometimes is referred to as the *underlay*. Operators can realize optimal network efficiency by dynamically assigning mobile terminals to either the overlay or underlay layers depending on signal strength and quality. Different infrastructure vendors employ different algorithms and different measurements (such as C/I versus power or timing advance) for managing these channel assignments. A variety of vendors have implemented solutions based on this (or similar) approaches. Ericsson refers to their implementation as "Multiple Reuse Pattern," Lucent as "Variable Interference Planning," Motorola as "Concentric Cells," Nokia as "Intelligent Underlay Overlay" and "Intelligent Frequency Hopping," and Nortel as "Cell Tiering." Another variable is whether a mobile terminal supports AMR. These terminals can be more readily assigned to the underlay layer while non-AMR mobile terminals can be assigned to the overlay.

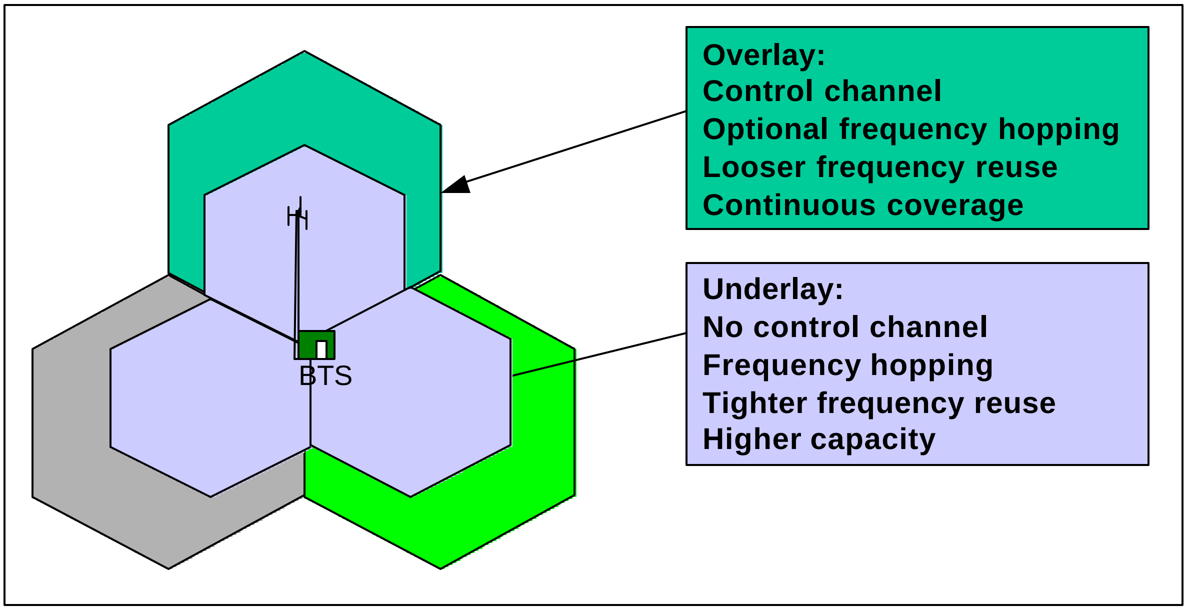


Figure 4: Re-use partitioning

Discontinuous Transmission

Any method that decreases interference can help increase capacity. One such method is called *discontinuous transmission* (DTX). In an average conversation, each user speaks less than 50% of the time. Normally, transmission would occur even when a person is not talking. The basis of DTX is to minimize transmission during these times, thus significantly reducing co-channel interference in nearby cells using those same frequencies. The detection of speech is referred to as *voice activity detection* (VAD).

DTX does not eliminate transmission altogether during non-speech times, as the intermittent silence would be distracting. Instead, a smaller number of frames are sent containing "comfort noise" that blends with the speech. Measurements show that DTX can increase capacity by 30%.

DTX can operate in both the downlink and the uplink. In the uplink, it has the further benefit of reducing overall power consumption of the mobile terminal, which improves battery life.

Dynamic Power Control

By reducing transmission power of both the base station and the mobile terminal to the minimum necessary, interference in neighboring cells can be minimized. GSM communicates the power control information in the slow associated control channel (SACCH), which it sends once every .48 seconds. Measurements show that power control can increase capacity by 50% in the hopping layer.

Half-rate Channel Mode

In half-rate channel mode, time slots multiplex two voice conversations. Though half rate is not possible with the current EFR codec, the AMR codec has six different modes for half-rate operation. In higher carrier/interference conditions, voice quality in half-rate mode is close to full-rate operation. While it might seem that packing sixteen users into each radio channel versus eight would increase voice capacity, this is actually not the case due to a higher C/I requirement. In other words, an operator can either have half-rate users in a smaller number of radio channels or the same number of full-rate users in a greater number of radio channels. However, because these full-rate users are in channels with a lower C/I requirement, the network can allocate more of them through higher fractional loads.

The area where half-rate channel mode *does* provide a benefit is that a lower number of radios can support the same number of active voice users. Analysis shows that an operator can reduce the number of radios in base transceiving stations between 28% and 39%. The half-rate codec modes do not include the highest-quality speech modes available in full-rate (12.2 Kbit/s and 10.2 Kbit/s) channels. The voice quality can therefore be lower in half-rate channel mode, though the actual amount is only a minor 0.2 in Mean Opinion Score (MOS) measurements in the very good speech quality areas. Half-rate mode is also planned for EDGE networks discussed below.

Dynamic Frequency and Channel Allocation

Another important and new method is called Dynamic Frequency and Channel Allocation (DFCA), which dynamically assigns radio channels in the optimum manner based on the current radio environment. This approach requires a synchronized GSM network. For all new connections (or handovers), the network chooses a channel (frequency and time slot) with the most suitable C/I ratio. The idea is to provide the C/I required for the new connection without increasing the interference beyond an acceptable level for existing connections.

This results in an extremely efficient use of the spectrum regardless of the geographical distribution of the mobile terminals. This method, which can further increase capacity by up to 40 to 50%[6], supercedes previous automated planning approaches, which are more static in comparison. Using DFCA with AMR, GSM capacity in 10 MHz of spectrum reaches 170 Erlangs per sector, compared to 156 Erlangs for CDMA2000 1XRTT, as discussed in the section "Technology Comparisons."

Future Enhancements

One extremely promising approach for increasing capacity is to decrease C/I requirements by determining what portion of a signal is interference. 3GPP Release 6 activities are considering methods referred to as Single Antenna Interference Cancellation (SAIC). One specific method is called joint demodulation, which relies on detecting the training sequences (additional bits available in each TDMA frame) of both the wanted and the interfering signals, and using them to estimate the channel. This method yields maximum benefit in a synchronous network where all base stations in a coverage area are synchronized with each other. SAIC could significantly increase GSM capacity. Preliminary estimates show gains of 60% to 100% when there is control of the interference distribution, as provided by Dynamic Frequency and Channel Allocation.

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[6] If AMR and re-use partitioning are already in use, additional gains from DFCA are 20 to 25%.

Other methods are under consideration that could help GSM and other mobile standards (e.g., smart antennas, mobile receive diversity) and not all are described in this paper. But having now described eight significant means of improving GSM spectral efficiency, the scope of innovation with GSM should be obvious.

Technology Comparisons

We now quantify the capacity improvements in GSM and compare GSM against other networks, including analog (AMPS), TDMA, UMTS, and CDMA2000.

First, in Figure 5, we look at GSM voice capacity. Here we need to specify whether the GSM broadcast control channel (BCCH) is in use or not. This might be relevant for a GSM operator considering a 1XRTT overlay. In that scenario, the operator already has the BCCH, and would be allocating additional spectrum either for 1XRTT or a GSM hopping layer. In each column pair, the first column includes the BCCH and the second column does not.

Referring to Figure 5, the first column pair shows basic GSM voice capacity using the EFR codec, frequency hopping, power control, DTX, and re-use partitioning techniques. The second column pair shows the capacity improvements with the addition of AMR. This figure clearly shows the effectiveness of AMR in increasing GSM capacity.[7]

[7] Additional assumptions include 4/12 re-use for BCCH channels and 1/1 re-use on the hopping layer, a fractional frequency load of 25% with EFR and 52% with AMR, a grade of service with 2% blocking, 95% of the speech samples have a FER better than 2.5%, and there is an SDCCH every fourth carrier. The average GSM AMR coder rate is greater than 8 Kbit/s, making it comparable to 1XRTT EVRC 8 Kbit/s. A 200 kHz guard band is also assumed.

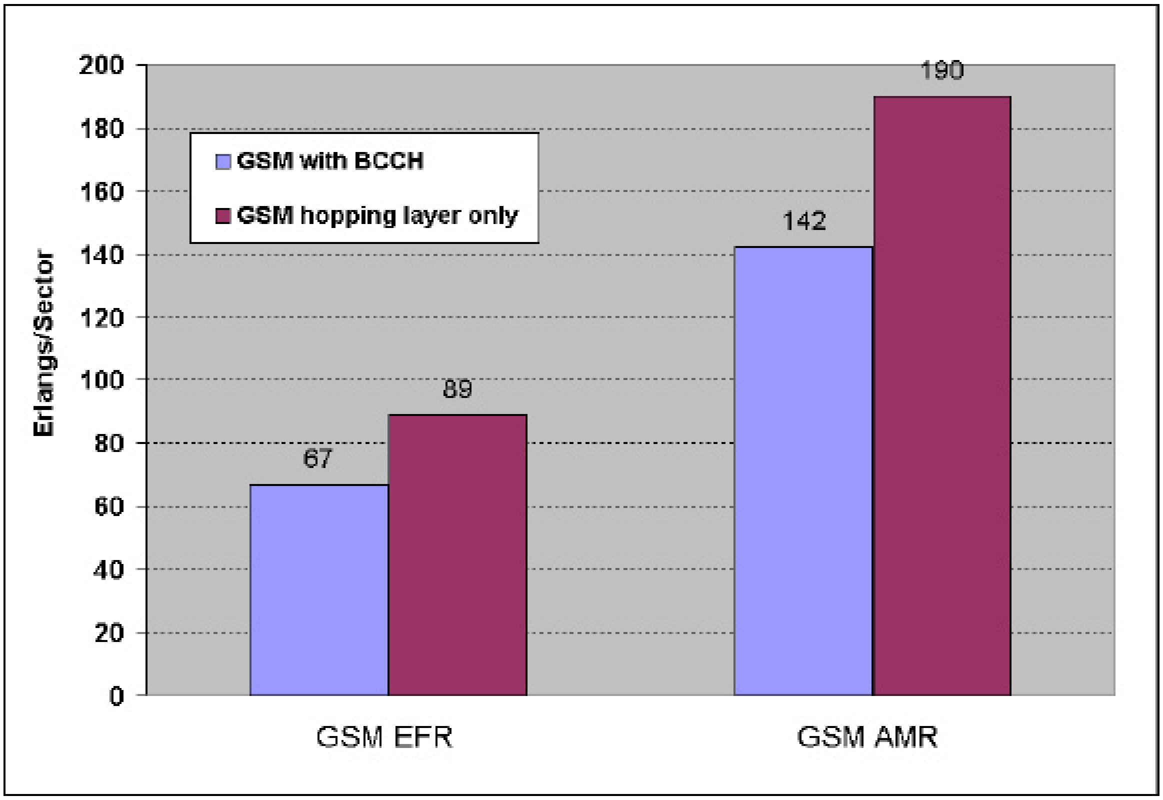

Figure 5: GSM capacity improvements in 10 MHz spectrum with AMR

Next we compare GSM against analog, TDMA, and CDMA2000 1XRTT, as shown in Figure 6.

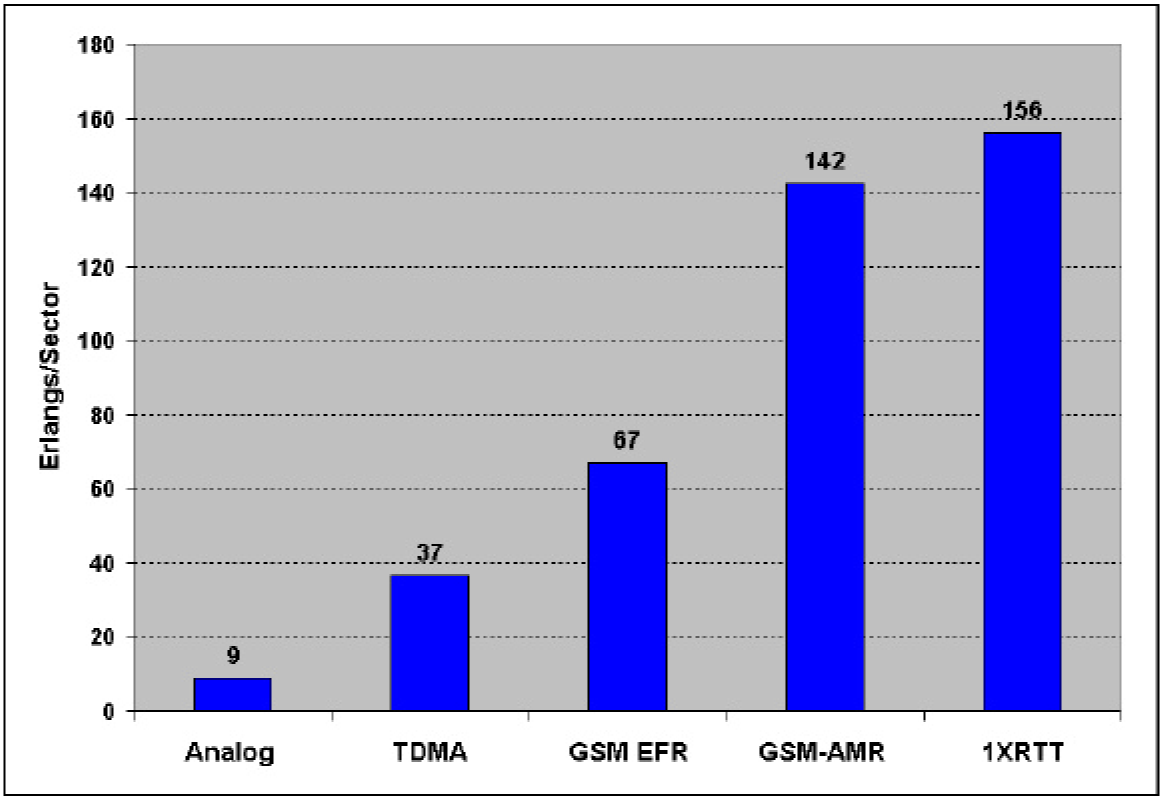

Figure 6: Comparison of analog, TDMA, GSM, and 1XRTT (in 10 MHz)

These values assume the BCCH in all GSM figures. GSM and 1XRTT data is based on infrastructure-vendor-agreed numbers that can be met in a realistic environment.[8] The first obvious conclusion is that GSM offers far higher voice capacity than TDMA, and up to a sixteen-times improvement over analog using available equipment. The second conclusion is that GSM voice capacity is nearly equivalent to CDMA2000 1XRTT. With the addition of DFCA, which can provide an additional gain of 20 to 25%, GSM voice capacity reaches 170 Erlangs and exceeds 1XRTT. With forthcoming technologies such as SAIC, GSM could become the most advanced technology available for speech services.

The final comparison is between GSM, 1XRTT and UMTS as shown in Figure 7.

[8] Assumptions include seven 1XRTT carriers per 10 MHz and 95% of the speech samples have a FER better than 2.5%.

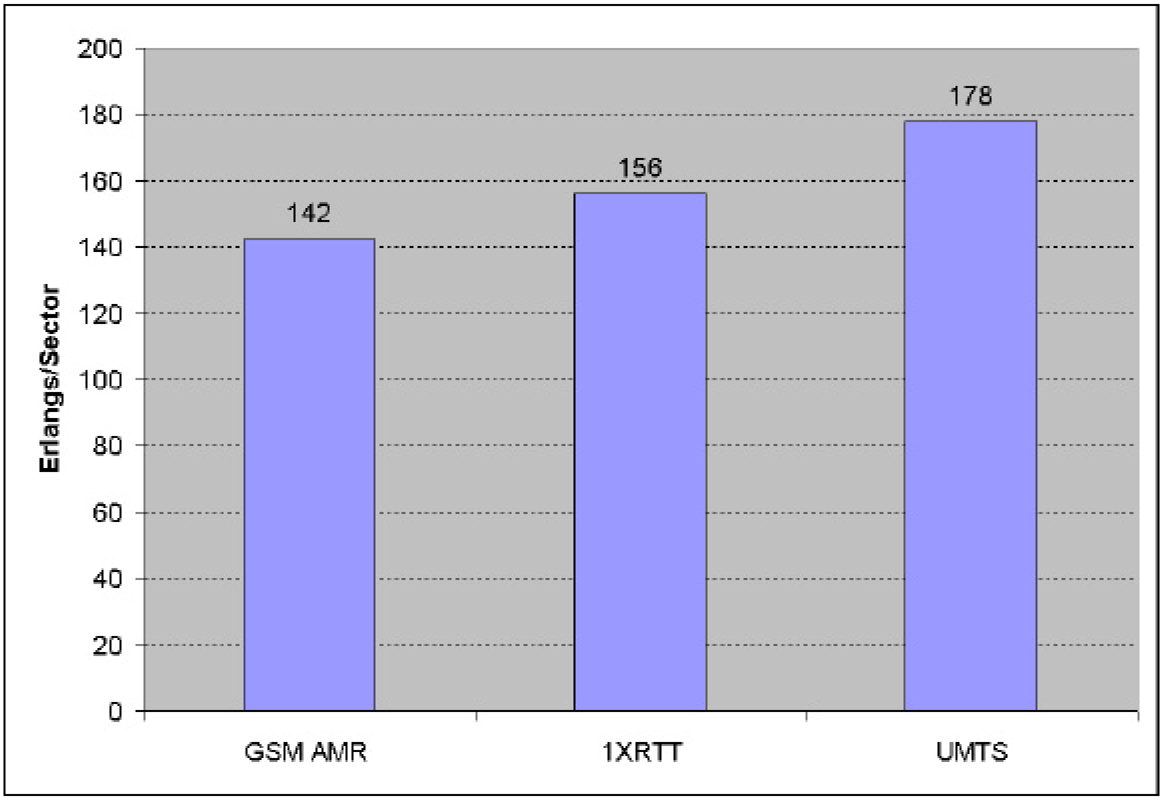

Figure 7: Comparison of GSM, 1XRTT, and UMTS (in 10 MHz)

This figure depicts UMTS using the AMR 7.95 Kbit/s codec rate.[9] What one can observe is that UMTS delivers higher voice capacity than GSM or 1XRTT. One planned enhancement for 1XRTT is the Selective Mode Vocoder (SMV). However, similar source adaptation methods can be applied to GSM and UMTS for comparable capacity gains (about 20%).

We discuss the evolution of GSM to EDGE and UMTS in the next section.

GSM Evolution to EDGE and UMTS

Operators deploying GSM today have a powerful and capable technology that not only matches competing technologies, but also is a global standard with deployment in nearly every country worldwide. GSM is a constantly evolving technology, and the next major milestones are Enhanced Data Rates for GSM Evolution (EDGE) and the Universal Mobile Telecommunications System (UMTS). Both bring new capabilities for voice.

Of particular benefit to operators is that the same core network will support all of these radio access networks, allowing GSM operators to selectively deploy EDGE and UMTS as needed or desired for additional capabilities and services. This is referred to as the UMTS Multi-radio network.

[9] All networks are based on effective speech coding of 8 Kbit/s. UMTS capacity assumes transmit diversity.

EDGE Voice Capabilities

EDGE is a radio technology that works in conjunction with GSM and GPRS networks to deliver higher data rates. These higher rates can theoretically be as high as 473 Kbit/s in the network, with peak device rates of 236 Kbit/s and typical user rates of 80 Kbit/s to 130 Kbit/s, a tripling of GPRS data rates.[10]. Though the principal motivation for operators to deploy EDGE is higher data rates, EDGE will also enable voice capacity improvements. This work belongs to the GSM/EDGE Radio Access Network (GERAN), part of Release 5 of the 3GPP specifications.

EDGE works by dynamically selecting channel coding (the amount of error correction relative to payload) and modulation, delivering the fastest throughput possible based on current radio conditions. Whereas GSM and GPRS use a modulation scheme called Gaussian Minimum Shift Keying (GMSK), EDGE uses both GMSK and a modulation called Octagonal Phase Shift Keying (8-PSK). 8-PSK can communicate three bits of information in each radio symbol (modulation of the carrier waveform) whereas GMSK can communicate only one bit. To support EDGE, base-station radio equipment must be capable of 8-PSK. Most of the new networks being deployed today (including those by North American carriers) are EDGE capable. 8-PSK modulation can also be applied to voice channels.

Since 8-PSK delivers a higher throughput of bits in each time slot, more bits can be assigned for error correction, thus making AMR half-rate mode (two users multiplexed into each time slot) much more feasible. 8-PSK is expected to boost voice capacity by 15 to 20%[11]. The higher bit rate will also enable all the AMR modes, including the 10.2 and 12.2 Kbit/s coders that otherwise are not available with GMSK modulation.

Another important technique that improves voice capacity in Release 5 is enhanced power control (EPC). While GSM today can change power levels every 480 msec (in each slow associated control channel block), with EPC GSM can adjust the power level every 120 msec (every SACCH burst). For faster moving mobiles, EPC can improve network capacity by 20%.

UMTS Voice Capabilities

The Universal Mobile Telecommunications System is a powerful cellular technology that carriers are beginning to deploy in markets worldwide. UMTS offers features that make it a particularly compelling upgrade from GSM. First, UMTS supports high-speed data of up to 10 Mbit/s with the new High Speed Downlink Packet Access (HSDPA). Second, UMTS does not involve a complete replacement of user equipment and infrastructure. Carriers can deploy UMTS progressively, with both GSM and UMTS networks operating in the same area. New multimode mobile terminals will support GSM, GPRS, EDGE, and UMTS. GSM will provide coverage in broad continuous areas. UMTS can provide additional voice capacity and advanced data services initially in the densest population areas and eventually in all coverage areas.

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[10] Assumes four time slots used for data for both GPRS and EDGE operation.

[11] These gains are not currently cumulative with SAIC.

UMTS uses a wideband CDMA radio interface. This is a direct-sequence, spread-spectrum radio system like CDMA2000, but it operates in wider radio channels of 5 MHz (compared to 1.25 MHz for CDMA2000). These wider radio channels provide significant advantages, including greater frequency diversity and flexible management of capacity between voice and data services.

From a pure voice capacity point of view, this paper has already described how GSM in combination with a variety of sophisticated mechanisms has equivalent capacity to next-generation cellular systems. But we have not yet discussed one of the largest advantages of UMTS: the flexible management of overall capacity between multiple types of services. To understand this benefit, we need to examine the UMTS radio interface.

UMTS, though considered a code-division system, is actually a combination of a code-division multiple access and time-division multiple access. UMTS allocates different codes for different channels, whether for voice or data, but UMTS adjusts the amount of capacity, or code space, of each channel every 10 msec. See Figure 8.

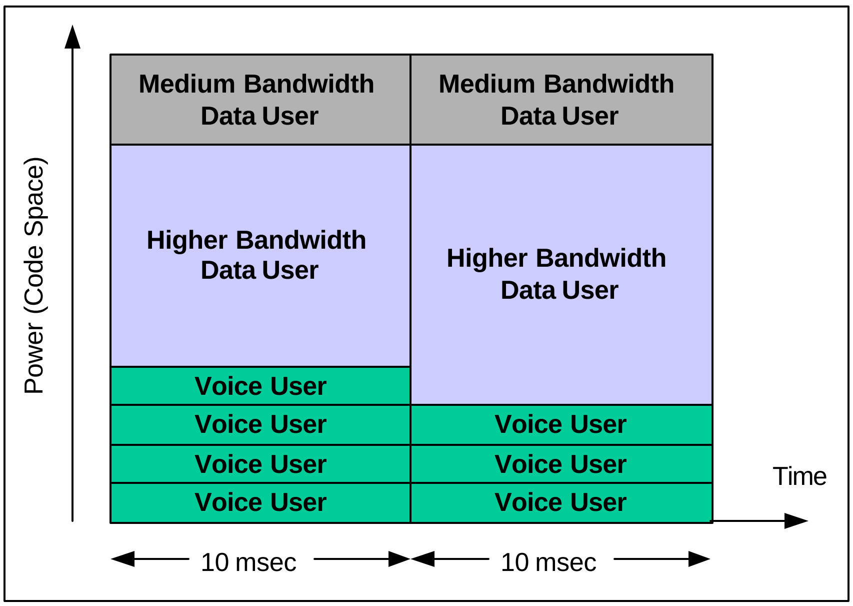

Figure 8: UMTS (W-CDMA) bandwidth management

Voice users require a relatively constant amount of bandwidth, but demand from data users can vary tremendously moment by moment. UMTS can manage this demand, allocating capacity where and when needed. UMTS controls capacity of each traffic channel by varying the amount of spreading. A lower amount of spreading translates to a higher capacity channel, but with fewer channels that can be simultaneously active. The net result is high peak throughput speeds as well as a fair sharing of capacity among many users.

Relative to CDMA2000, UMTS enjoys a number of technical advantages. First it uses a higher chip rate of 3.84 Mchips/s (which controls the spreading), compared to 1.2288 Mchips/s. This provides better multi-path diversity, translating to improved coverage.

Another advantage is that UMTS can share high-speed data services and voice services in the same radio channel, whereas CDMA2000 requires a separate radio channel for high-speed data using 1X-EVDO (1X evolution data only) technology. This results in significant capacity underutilization because a lightly loaded data carrier cannot be used for voice and a lightly loaded voice carrier cannot be used for high-speed data. UMTS also has a sophisticated quality-of-service architecture not currently available in CDMA2000, which allows applications such as multimedia and voice over IP.

Finally, with respect to voice capacity, UMTS is an evolving standard. Much of the ingenuity that vendors have used to improve GSM will also be applied to UMTS. For instance, Release 5 of the UMTS specifications increases peak data transmission rates from 2 Mbit/s to 10 Mbit/s. Similar innovations can be expected with respect to voice capacity as the technology matures.

Summary and Conclusion

This paper has presented the case for the spectral efficiency of GSM, demonstrating innovative new methods for increasing voice capacity. As a result, GSM capacity meets or exceeds competitive technologies such as CDMA2000 1XRTT. GSM uses techniques as advanced and as sophisticated as 1XRTT, and delivers comparable or better capabilities and services.

This paper has categorized techniques for improving GSM spectral efficiency as either mature, emerging, or still in development. Of these, frequency hopping is the most powerful mature method, the AMR codec is the most powerful emerging method, and SAIC is the most promising method in development. This paper has described eight different methods by which GSM improves voice capacity, summarized below:

- The AMR codec dynamically allocates available throughput to either voice coding or error control to provide the best voice quality under current radio conditions, and to provide reliable voice operation when the network employs tighter frequency re-use and higher fractional loading.
- Frequency hopping mitigates interference by increasing frequency diversity, and enables tighter frequency re-use, as tight as N=1, where each frequency is re-used in every sector.
- Re-use partitioning exploits frequency hopping to create an overlay network with continuous coverage and an underlay network with tighter re-use with increased capacity. The network can intelligently assign calls to either the overlay or underlay network.
- Discontinuous transmission further improves capacity by limiting transmissions to times of active speech.
- Dynamic power control limits transmission power levels to the minimum necessary.
- With Dynamic Frequency and Channel Allocation, the network monitors the instantaneous radio environment, and dynamically allocates frequencies and time slots to assign the most suitable C/I while minimizing interference to existing connections.
- Half-rate mode multiplexes two users in each time slot to increase network efficiency, and becomes even more effective with 8-PSK modulation used in EDGE.

Finally, this paper discussed the evolution beyond GSM. With Single Antenna Interference Cancellation, GSM has the potential of becoming the most spectrally efficient technology for speech services. Beyond this, UMTS provides operators additional capacity, quality-of-service mechanisms, and flexibility in managing resources between voice and data services, while the

UMTS Multi-radio network allows simultaneous operation of GSM, EDGE, and UMTS radio access technologies.

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