Introduction

The standardization of the first releases of third-generation cellular systems was finalized in ETSI/3GPP Release 99. Two major systems in this standard are the universal mobile telecommunications system (UMTS) and the global system for mobile communication/enhanced data rates for global evolution (GSM/EDGE).

UMTS has been tailored to offer third-generation services—that is, high user data rates for real-time as well as traditional non-real-time applications. The tailoring comprises both the UMTS core network and the WCDMA-based UMTS terrestrial radio access network.\(^1\)

UMTS is currently intended for use in the designated UMTS/IMT-2000 spectrum (and possibly 1,900 MHz), whereas the aim of GSM/EDGE is to evolve GSM and TDMA to provide coverage for third-generation services in existing spectrum bands (700, 850, 900, 1,800 and 1,900 MHz), and to complement UTRAN as a fully capable radio access network that can be connected to the UMTS core network.

By introducing high-level modulation in the GSM radio interface, it is expected that EDGE will successfully manage to provide support for high data rates. The next step in the GSM/EDGE evolution is to provide support for conversational and streaming service classes (real-time services). A driver of this evolution is the shift within the telecommunications world from circuit switching to packet switching. This shift is valid for traditional non-real-time data services, such as e-mail and Web browsing, as well as for real-time services, such as video conferencing and voice over IP (VoIP).

The second-generation packet-switched core network (which was defined for general packet radio service, GPRS) and the current GSM/EDGE radio access network must each be modified to support real-time services. This includes adopting the Iu interface to the third-generation UMTS core network. Doing so simplifies the alignment of services that will be provided in UMTS,
and enables connection to the same third-
generation core network.

In the 3GPP standardization, this next
phase of evolution is called GSM/EDGE
radio access network or GERAN (Box B).
The two main objectives of GERAN are:

- the alignment of GSM/EDGE and UMTS
  services—this mainly relates to providing
  conversational and streaming service
  classes; and

- the ability to interface with the third-
generation UMTS core network over the
  Iu interface used in UTRAN.

In addition, GERAN will enhance the
performance of existing services.

The first phase of the GSM/EDGE con-
cept has already been presented and evalu-
ated.\(^2\) This article focuses on the subsequent
evolution, describing the overall challenges
of supporting real-time services in GERAN.

**Why GERAN?**

The motivation behind the standardization
activity for evolving GSM/EDGE toward,
and aligning it with, UMTS can perhaps
best be described in terms of target group—
that is, in terms of operators, vendors, and
end-users.

For mobile operators and vendors, the
prime driving forces are directly related to
reduced operating costs and increased rev-
enues. Operators want to use existing (rela-
tively cheap) GSM frequency bands as a
complement to the newly acquired UMTS
bands. And because GERAN offers the same
third-generation bearer services as those de-
dined for UMTS, any new investments that
operators make in equipment for the GSM
bands will be secured for the foreseeable fu-
ture. Operators might also look forward to
closer cooperation and integration between
their different networks (GSM/EDGE and
UMTS). This can partially be achieved
through an interface between the base sta-
tion controller (BSC) and radio network con-
troller (RNC).

Vendors want the solutions for imple-
menting GSM/EDGE to facilitate conver-
gence with the development and manufac-
turing processes for UMTS. Happily, the
prerequisites for achieving this objective are
provided through the GERAN enhance-
ments, whose prime goal is alignment with
UMTS. This requirement also guaran-
tees long-term revenue streams from
GSM/EDGE equipment.

Obviously, end-users must also perceive
the benefits of the new technology, for ex-
ample, by being able to seamlessly roam be-
tween different radio access technologies
(GERAN or UTRAN) to access third-
generation UMTS services, such as real-time
IP multimedia. And because application de-
velopers will now have a uniform environ-
ment for creating new and advanced third-
generation services, these services can thus
be made available to end-users in a timely
manner.

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**BOX B, STANDARDIZATION OF GERAN IN 3GPP**

ETSI (STC SMG2) standardized the GSM radio
interface and made enhancements to EDGE. The
introduction of the Iu mode motivated a
closer coupling to 3GPP, and consequently, in
the summer of 2000, ETSI/SMG2 was trans-
fected to the 3GPP Technical Specification
Group (TSG) GERAN, and all existing ETSI spec-
fications were adopted as 3GPP specifications
under new numbers. Release 99 thus became
the last release of the GSM/EDGE standard
delivered by ETSI. TSG GERAN has subgroups
for radio aspects, protocol aspects, BTS confor-
mance tests, and mobile station confor-
mance tests (Figure 1).

The term GERAN, which is short for
GSM/EDGE radio access network, is used in
standardization effective from Release 99. How-
ever, the term is more frequently associated
with the adoption of the UMTS architecture for the
GSM/EDGE; that is, the adoption of the Iu inter-
face, which is to be included in a later release.

After Release 99, all subsequent releases will
not be formed as yearly releases, but rather as
functional releases at regular intervals. Accord-
ingly, the coming releases will be called Release 4
(Rel-4), Release 5 (Rel-5), and so on.

While GERAN Rel-4 introduced only minor
enhancements to the standard, major enhance-
ment packages are expected to come with
Rel-5 and Rel-6—GERAN Rel-5 will introduce
- the UMTS architecture (that is, the Iu inter-
face and protocol architecture); and
- all other enhancements related to the radio
interface (for example, support for the con-
versational service class and IP multimedia).

In terms of contributions and participation,
Ericsson is one of the leading companies in TSG
GERAN.
GERAN support for UMTS
QoS classes in packet-switched domain
Second-generation radio access technology brought mobile telephony to a broad market. By contrast, third-generation radio access technology will extend beyond basic telephony: a common, IP-based service platform will offer mobile users an abundance of real-time and non-real time (traditional data) services.

Typical services with real-time requirements are voice and video, as well as delay-sensitive applications, such as traffic-signaling systems, remote sensing, and systems that provide interactive access to World Wide Web (WWW) servers.

The challenge is to implement end-to-end services based on the Internet protocol (IP). The main benefit of running IP end-to-end—including over the air interface—is service flexibility. Indeed, flexibility more or less eliminates dependencies between applications and underlying networks, for example, access networks. To date, cellular access networks have been optimized in terms of voice quality and spectrum efficiency for circuit-switched voice applications. However, for services such as IP multimedia, which includes voice, the main challenge is to retain comparable quality and spectrum efficiency without decreasing service flexibility. Today, for example, we can suffer considerable protocol overhead when we bridge the air interface with real-time protocol (RTP), user datagram protocol (UDP) or IP packets (which carry media frames).

To achieve spectrum efficiency, we can instead characterize different packet-data streams in terms of bandwidth and delay requirements. Characterization of this kind is useful when implementing admission-access algorithms that accommodate multiple user data streams in available spectrum. Different methods of limiting data (such as header compression and session signaling compression) must also be applied to obtain adequate spectrum efficiency.

QoS classes in UMTS and GSM/EDGE
Because frequency spectrum is a sparse resource, we readily see the benefit of classifying traffic to guarantee system capacity and quality of service (QoS). By differentiating traffic flows in the network, four application-related QoS classes have been defined within UMTS and GSM/EDGE:

- The conversational service class is used for real-time services, such as ordinary telephony voice—that is, IP-telephony and video-conferencing. The vital characteristics of this class are low transmission delay and preserved time relationships, or low-delay variation, in the traffic flow.
- The streaming service class applies to real-time audio and video-streaming applications. In contrast to the conversational class, this category comprises one-way transport.
- Typical applications associated with the interactive service class are WWW browsing and telnet. The fundamental characteristic of this service class is a request-response pattern. Consequently, the round-trip delay is an important factor.
- The background service class is used for best-effort traffic. Examples of services in this class are electronic mail (e-mail), short message service (SMS), and file transfer. In this service class, the requirements that apply to transfer delay are less stringent.

These four QoS classes are supported in GERAN and UTRAN by adequate radio access bearers.

Radio access bearer realizations in GERAN
Radio access bearers (RAB) provide bearer services through the radio access network. Each radio access bearer is associated with a set of attributes that

- specifies the required quality; and
- supplies information on the characteristics of the traffic flow.

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<th>BOX C, TERMINOLOGY</th>
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<td>A-interface</td>
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<td>EDGE phase 2</td>
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Some examples of attributes are service class, maximum bit rate, service data unit (SDU) loss rate, residual bit error ratio, transfer delay, and guaranteed bandwidth. This information is essential for providing a connection with good quality through the radio access network and for using spectrum efficiently. The different QoS classes specified for UMTS are characterized by specific value ranges for different RAB attributes.

The radio access bearer services provided by GERAN will be aligned with those supported by UTRAN through adapting and supporting the complete UMTS QoS model. The UMTS service concept is thus reused and each QoS class can be supported independently of the radio access network.

GERAN currently supports interactive and background service classes over the Gb interface (Release 99). Further enhancements to the cell re-selection procedure, which will shorten the interruptions during cell changes, will enable support for the streaming service class in Release 4 (Rel-4). Finally, all four service classes specified for UMTS will be supported by GERAN Release 5 (Rel-5) when operating in the Iu mode. That is, in the 3GPP standardization (Box B), it was decided that the conversational service class will solely be supported in Iu mode. Consequently, handover mechanisms with pre-allocation for resources for the packet-switched domain are being introduced in GERAN Rel-5 specifically for the Iu mode of operation.

**Bearers for voice service in GERAN**

The GERAN standard supports all classical GSM circuit-switched voice services—full-rate (FR), half-rate (HR), enhanced full-rate (EFR), and adaptive multirate (AMR), as well as narrowband HR and FR. As in GSM Release 98, all these services can be realized over the A-interface using Gaussian minimum-shift keying (GMSK) modulation on the physical layer. GERAN Rel-5 also discusses the introduction of AMR half-rate services using eight-phase-shift keying (8-PSK) modulation as an enhancement to existing AMR half-rate voice on the physical layer. GERAN will also support a wideband AMR voice codec.

**GERAN system architecture**

One might think that providing support for packet-based real-time services and the adoption of the UMTS QoS architecture would require that changes should be made to the second-generation GPRS core network. However, an alternative solution is to connect GERAN to the third-generation UMTS core network, which supports real-time services and the UMTS QoS architecture. This approach employs a common core network for UTRAN and GERAN over a common interface.

To connect to the third-generation UMTS core network, GERAN must use the Iu interface (3GPP Release 5 of the specification). This interface can be seen as comprising two parts: the Iu-ps connects to the packet-switched domain of the core network, whereas the Iu-cs connects to the circuit-switched domain. Note: the two parts of the Iu interface share the control plane (the radio access network application part, RANAP, protocol) and have similar user planes.

Figure 2 shows that GERAN also connects to second-generation core network nodes. The Gb interface and the connection to the serving GPRS support node (SGSN) are needed for support of Rel-4 (and earlier) terminals. The Iu-ps interface could not be used for legacy terminals, since for the Iu-ps interface, the functional split between the radio access network (RAN) and the core network differs from that for the Gb inter-
The preferred solution is thus to support legacy terminals over the A/Gb interfaces.

The air interface between the mobile terminal (also called mobile station, MS) and GERAN, called the Um interface, is based in part on the radio link interface of Release 99. However, several enhancements being specified on different radio link protocol layers will provide adequate radio bearers for real-time packet services. Examples of these enhancements are support for handover for the packet-switched domain, separation of the user and control planes, and transparent modes in radio link protocol layers.

Functional split

Figure 2 shows the overall GERAN reference architecture. At a high level, the GERAN base station subsystem (BSS) consists of the base transceiver system (BTS) and the BSC. Unlike in UTRAN, the functional split of the GERAN BSS is not specified in the standard. In GERAN, the radio link controller (RLC) and medium access control (MAC) protocol can be placed in any BSS node or even in the core network. This architectural freedom remains in GERAN Rel-5. However, at a functional level, there are some important restrictions that follow from adopting the Iu interface in the core network. For instance, user data must be ciphered on the RLC or MAC protocol layers. Additionally, when operating in circuit-switched mode over the Iu interface, the voice transcoders must be located in the core network (Figure 3).

Modes of operation

As indicated in Figure 2, GERAN can be connected to

- a second-generation core network via the A- and Gb interfaces; and
- a third-generation core network via the Iu-ps (for the packet-switched domain) and Iu-cs (for the circuit-switched domain) interfaces.

According to the standard, a mobile terminal may not connect to the second- and third-generation core networks simultaneously. It must operate in either A/Gb mode or Iu mode. This restriction was stipulated to avoid very complex coordination of the two modes of operation in a multiservice scenario. The GERAN, on the other hand, can provide simultaneous services from the second- and third-generation core networks to different mobile terminals. In addition, GERAN is required to multiplex traffic to and from mobile terminals that operate in different modes onto the same physical channel over the air interface: timeslot or half timeslot.

The Iur-g interface

An important new feature is the option to have two BSCs, as well as a BSC and a UTRAN RNC, connected through the Iur-g interface (Figure 2). The main advantages of the interface, which has been de-
signed to carry control signaling only, are
• greater alignment with the UMTS archi-
tecture (which splits the core network
from the radio access network) by allow-
ing the definition of RAN internal regist-
tration areas. These can span over multi-
ple cells that belong to different GERAN
BSCs, and UTRAN RNCs. The use of
registration areas reduces mobility man-
gement signaling to the core network;
• improved performance of the cell-
reselection procedures within GERAN; and
• support for the serving RNC or serving
BSC relocation procedures as defined in
UTRAN. This improves the performance
of the cell re-selection procedure between
UTRAN and GERAN.
It might also be possible to further enhance
the Iur-g interface to achieve traffic steering
and load distribution between GSM/EDGE
and UMTS.

Protocol models
When introducing the Iu interface in
GERAN, the designers endeavored to
interject as few changes as possible to the
protocols that relate to this interface as well
as to the core network nodes and to
UTRAN. Consequently, they adopted the
associated Iu protocols—mainly the
RANAP protocol. However, on the radio
side, the introduction of the Iu interface
meant that the protocol stacks of
GSM/GPRS had to be modified (see Figures
4 and 5, which show the protocols influ-
enced by the Iu and A/Gb interfaces).
The most noticeable difference in the user
plane of the packet-switching mode is that
the packet data convergence protocol
(PDCP) is used as a radio link layer proto-
col for operation over the Iu-ps interface (in-
stead of the SNDCP/LLC protocols, which
are used for operation over the Gb interface),
and that a transparent RLC mode has been
introduced (Figure 4).
In the user plane of the circuit-
switching mode, the Iu-cs interface partly
uses the same protocol stack as that of the
Iu-ps interface, but operates with RLC and
MAC protocol layers in transparent mode.
The protocol stack for the A-interface has been
introduced (Figure 4).
In the control plane, the packet- and
circuit-switching channels are handled by
the radio resource control (RRC) protocol
when the Iu interface is used; the radio re-
source (RR) protocol of GSM/GPRS is
reused when the A/Gb interface is used. The
only exception is for common control chan-
nels (such as the broadcast control channel,
BCCH). In this case, the RR protocol is used
regardless of service (Figure 5).

Transport in GERAN
The transport options on the GERAN
interfaces comply with the UMTS standard.
As such, transport over the Iu and Iur-g
interfaces is not limited to what has been
specified to date (for example, asynchronous
transfer mode, ATM) but will also adopt
other options (such as IP) when they have
been standardized by 3GPP.

GERAN mobility management
As in UMTS, when the mobile terminal is
connected in GERAN Iu mode, all cell-level
mobility functionality resides in the radio
access network. This is a major departure
from GERAN operating in A/Gb mode, where cell-level mobility is handled by the core network. A feature of GERAN mobility management in Iu mode is that it allows mobile terminals in connected mode to use common registration areas for GERAN and UTRAN. This reduces signaling over the core network interface.

As explained above, one GERAN cell can connect to the core network via the A/Gb and Iu interfaces. In fact, one GERAN cell can simultaneously connect to second- and third-generation core networks, regardless of whether they are separate or have been integrated (Figures 6 and 7).

**GERAN physical layer**

GERAN must be able to multiplex, on the same physical channel, traffic from mobile terminals operating in Iu mode and A/Gb mode. Accordingly, the GERAN physical layer must remain backward compatible. However, several features that will enhance the operation of the physical layer are under discussion in standardization groups, including:

- enhanced power control;
- mobile allocation index offset (MAIO) hopping; and
- flexible layer-one concept.

**Enhanced power control**

GERAN Rel-5 will allow power control to operate on a 120 ms basis—as compared to the current 480 ms. This enhancement will increase capacity in multipath fading environments.

**MAIO hopping**

Ericsson has proposed an MAIO hopping algorithm that improves interference diversity. The idea is to improve the current frequency-hop algorithm (where mobile terminals in one cell hop together) to let mobile terminals hop with different offsets, thereby increasing the diversity in co-channel and adjacent-channel interference.

**Flexible layer-one concept**

A drawback of the current GSM/EDGE system is that layer one is relatively inflexible. Standards groups must agree on new schemes for optimized channel coding each time a new radio bearer service (such as for wideband AMR or VoIP) is proposed. Ericsson proposes a new, flexible layer-one concept, whose parameters, such as error protection, channel coding, puncturing, and interleaving, are controlled by the radio access network. This way, the introduction
of a new service will only require a new set of parameters.

GERAN and IP multimedia
When IP multimedia and VoIP were first discussed in the wireless community, vendors and operators quickly identified the need to reduce the large overhead constituted by the IP transport headers in order to make more efficient use of spectral assets. In 3GPP, UTRAN adopted the robust header compression (ROHC) standardized in IETF. This method allows for a reduction of the nominal header size of 40 bytes (IPv4) to an average of 1 byte. This is possible, since most of the elements in the IP header do not vary particularly often. However, if significant changes in the information in the IP header occur, a larger header might need to be transmitted. In this sense, the ROHC algorithm requires a flexible physical layer—to accommodate varying header size, the channel coding of the payload must change to keep the total number of transmitted bits constant.

Given the traditional relative inflexibility of the physical layer of GSM/GPRS, the method of complete header removal was instead discussed for GERAN. However, despite the obvious gain in spectrum efficiency, removing the IP header rules out the possibility of numerous desirable features, such as multiple synchronized flows to a single mobile terminal, and connecting the VoIP flow to an off-the-shelf multimedia application—for example, to a PC.

Conclusion
The continuous GSM/EDGE standardization in 3GPP offers a common evolution path for GSM and TDMA that provides a cost-effective means of providing third-generation services within existing GSM/TDMA frequency bands.

With Release 99 of the ETSI standard, circuit-switched voice and packet-switched services without strict delay requirements (such as Internet access for Web browsing and e-mail) can be efficiently supported with adequate radio bearers. And with the concept currently being standardized in 3GPP for Release 5, the GSM/EDGE radio access network (GERAN) will provide support for true third-generation wireless services. This includes support for all the service classes specified for UMTS, which in particular, includes support for the conversational service class with its real-time requirements. Furthermore, interfacing to the third-generation UMTS core network over the Iu interface yields greater alignment with UMTS.

REFERENCES

3 3GPP TS 43.051, “GSM/EDGE Radio Access Network; Overall Description - Stage 2.”