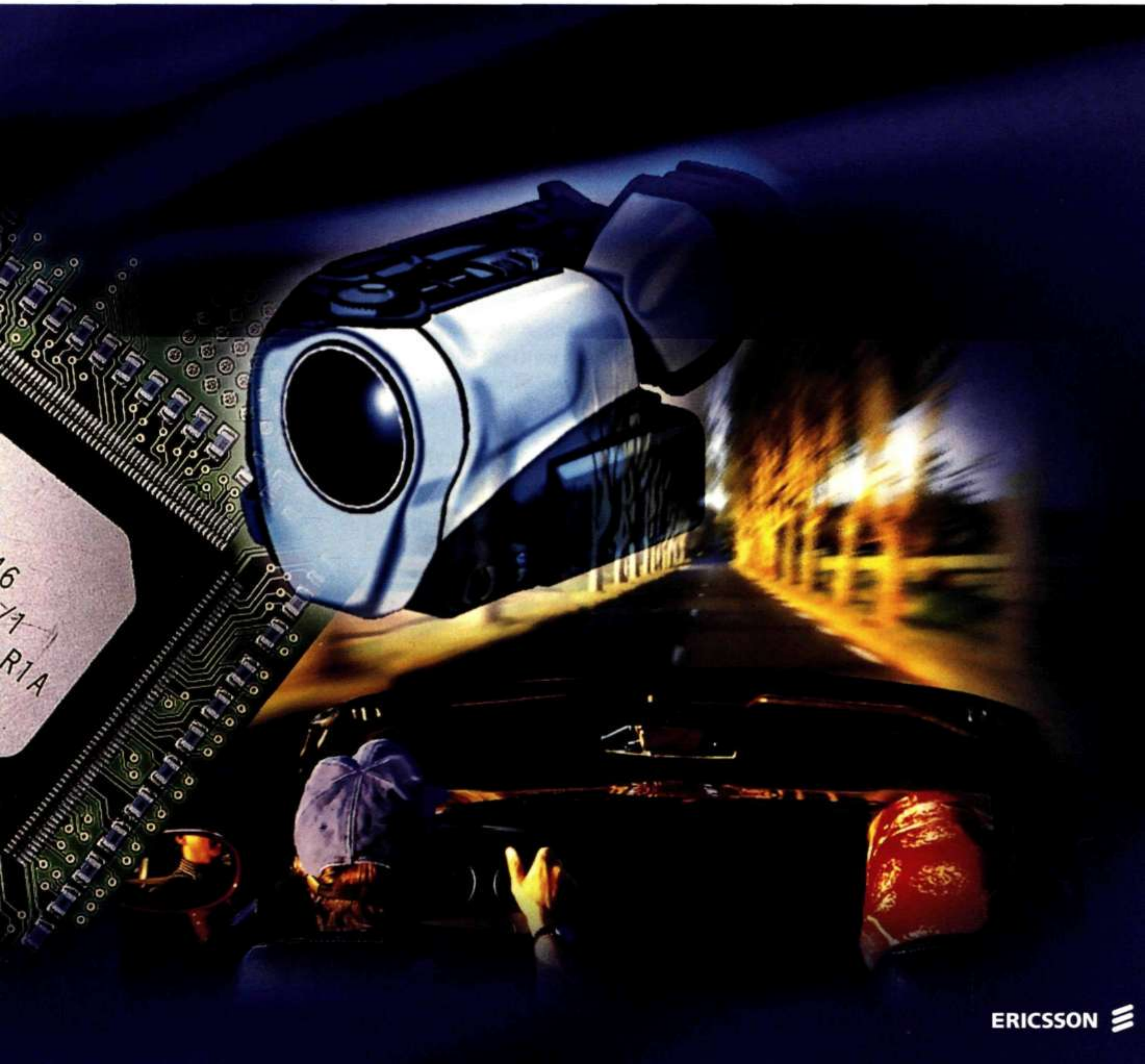
Ericsson
REVIEW
THE TELECOMMUNICATIONS TECHNOLOGY JOURNAL

NO 2, 1998



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REVIEW

The telecommunications technology journal

The purpose of Ericsson Review is to report on the research, development and production achievements made in telecommunications technology at Ericsson. The journal is published in English and Spanish and distributed quarterly to readers in more than 130 countries.

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Internet address:
http://www.ericsson.com/Review

Publisher: Håkan Jansson

Editor: Eric Peterson

Production manager: Eva Karlstein

Layout: Paues Media, Stockholm

Printer: Ljungföretagen, Örebro

ISSN: 0014-0171

Volume: 75, 1998

Cover: "Ericsson is leading the world into third-generation systems for wireless multimedia. And we are using the same recipe that brought us to where we are today—early development and close co-operation with operators who are demanding a cost-effective and spectrum-efficient technology. This will allow them to compete successfully into the 21st century. You know it's important to remember that 9 out of 10 of the world's largest operators are Ericsson customers."

SVEN-CHRISTER NILSSON
President and CEO

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From the editor

Eric Peterson

Next week I'm off to Anaheim, California for four days to attend the 45th annual conference of the Society for Technical Communication (STC) together with 2,000 or so other technical writers, editors, illustrators and the like.

As I think ahead to the conference, I can envision how much simpler and more convenient our lives will be when the WIN services described in this issue become commonplace. Imagine, for example, that as you register at a conference someone hands over a mobile phone along with your name badge and other conference materials! This way, conference delegates could easily stay in touch with one another and presenters and organizers could call directly for technical help when setting up for a presentation or (worse) when equipment breaks down in the middle of a presentation. Also, by extending the service to include access to hotel PBXs, delegates could make external calls and have them billed to their hotel account as well as have calls or voice mail forwarded to them from their hotel phones.

My schedule will be pretty tight next week, so I may not have time to take in the sights in a grand way, but I'm determined to squeeze in some gift shopping for my wife and daughters. Here again, access to WIN services and third-generation wireless communication could come to good use.

For example, let's assume that I decide to break away from the conference during lunch and dash over to a nearby specialty toy store. "I know exactly what I want," I think to myself, "so this shouldn't take long." However, upon entering the store, I am bowled over by a truly mind-boggling selection. And after 15 minutes of intensive browsing I have changed my mind half a dozen times, at least. "I wish my wife were here to advise me...wait! I've got it. I can hook up my multimedia PDA to my phone and e-mail her a quick note along with some digital images of the various alternatives I am considering for the girls!" And so I do. As I leave, I pick up a business card with the shop's e-mail and Web address on it. Later that afternoon, during a short break between sessions, I again hook up my PDA and phone and check for messages. My wife says to get the shiny, red gadget for our eldest girl, the blue-and-green gizmo for daughter number two, and the cuddly, yellow item for our youngest. "Perfect." I then initiate a secure connection with the store, transmit my credit card number and place my order, asking them to deliver it to the hotel where I am staying.

Of course, with the capabilities of third-generation wireless communication I might just as easily have called my wife from the shop and asked her to view the toys and advise me in real time.

What I have described is not science fiction. The technology already exists and you can read about it in this issue.

Other topics that you can read about include

- a brief introduction to six management-application areas of Ericsson's telecom-management products and services portfolio, which is based on Ericsson's Statement of Direction in providing telecom-management solutions for upper TMN layers;
- a description of mobile crosstalk control—a special algorithm that eliminates crosstalk echo and enhances speech quality in digital cellular networks;
- a look at the evolution of AXE core switching devices—with particular emphasis on an ingeniously simple concept of using generic device magazines for improving flexibility and drastically reducing time to customer.

I hope you enjoy this issue and find the articles to be timely, informative and pertinent.

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nications, and network management. He currently works as a strategic product manager at the Telecom Management Solutions business unit at Ericsson Radio Systems AB. He holds an MSc in mechanical engineering from the Linköping Institute of Techology.

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Toward third-generation wireless communication

Torbjörn Nilsson

The next significant development in wireless communication will consist of enhancements to the radio access that enable true multimedia services to be delivered at high bit rates.

New third-generation wideband systems will deliver bit rates up to 384 kbit/s for wide-area coverage, and 2 Mbit/s for indoor or fixed applications, maximizing the efficiency of available radio spectrum. Today's digital wireless networks and standards will also evolve to provide similar capabilities.

The author describes the market and technology perspective on third-generation wireless communication and introduces the wideband wireless system that Ericsson has been developing since the early 1990s.

Figure 1
Mobile telephony now a mass-market service.

When the cellular mobile phone arrived in the early 1980s, it marked a turning point in telecommunications. Adding radio access to the core telephone network invalidated the concept of a telephone connected at a fixed point in the network. Similarly, it meant we had to rethink the idea of a telephone number indicating a fixed geographical location.

The ubiquitous access had great appeal to business people, who made up the original market. As network capacities increased and costs fell, the mobile phone quickly became a mass-market commodity (Figure 1).

In January 1998, there were more than 207 million mobile phone users worldwide. And the expectation is that this number will grow to 830 million by the end of 2003 (Figure 2). In many countries today, the cost of mobile phone calls is falling toward the level of fixed phone calls.

First-generation analog wireless systems were followed by second-generation, digital technologies that delivered important benefits in three main areas. Digital wireless technologies

- support a much larger number of mobile subscribers within a given frequency allocation (that is, they increase capacity);
- provide superior security and voice quality;
- lay the foundation for the value-added services (including data) that will continue to be developed and enhanced into the next millennium.

The wireless terminal has the potential to become a generic platform for, or gateway to, the complete range of communication services; that is, voice, data, video and multimedia. And network operators recognize that future revenue streams in competitive and mature markets will not be generated solely from providing voice connections, but also from more sophisticated services.

Internet and convergence, new drivers for wireless multimedia

In enterprise networks and fixed telephone networks, data traffic (especially intranet and Internet usage) is growing faster than voice traffic. Industry experts believe that this trend will spill over into wireless communication (Figure 3).

The convergence of media and services is being stimulated by the evolution of technology and the use of Internet techniques. For example, the miniaturization of products and the digitization of media (includ-

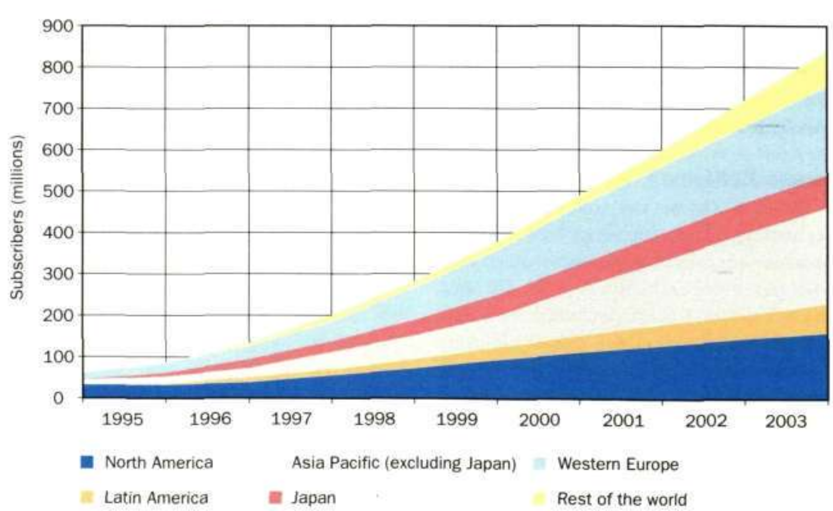


Figure 2
Forecast growth of cellular/PCS subscribers worldwide.

ing audio/video) make portable/pocketable multimedia devices a reality. Another important factor is that Internet standards are being used over PSTN/ISDN, cellular, CATV/broadcasting and paging networks.

Bandwidth at the air interface, however, is currently a limiting factor relative to the volume of information that can be transmitted between wireless mobile terminals and the network. This is because the aim of first- and second-generation mobile phone standards was mainly to support voice communication. Notwithstanding, many of today's mobile phones may be used for low-speed data applications, such as sending and receiving faxes and e-mail.

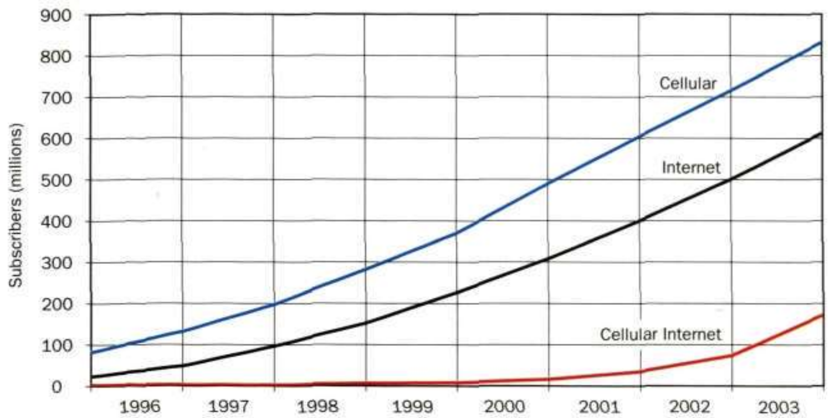


Figure 3
Forecast growth of Internet "subscribers" worldwide.

The third-generation vision and requirements

It is no easier to predict the future when planning for wireless communication than for any other area of telecommunications and information technology. Five years ago, who could have predicted what impact the Internet would have? The best we can do is to plan a wireless framework that is flexible, robust and powerful enough to cope with change.

In this light, the following services and applications embody the capabilities that are considered to be desirable in a third-generation wireless system.

- Full range of services—from narrowband voice up to wideband real-time multi media services. Voice traffic is expected to remain an important application and source of revenue.

Box A, Abbreviations

AAL2	ATM adaptation layer 2
ANSI	American National Standards Institute
ARIB	Association of Radio Industries and Broadcasting
ASCI	Advanced speech call items
ATM	Asynchronous transfer mode
BTS	Base transceiver station
CAMEL	Customized application for mobile enhanced logic
CATV	Cable TV
CDMA	Code-division multiple access
CDPD	Cellular digital packet data
CTS	Cordless telephone systems
D-AMPS	Digital advanced mobile phone service
EDGE	Enhanced data rates for GSM evolution
ETSI	European Telecommunications Standards Institute
GPRS	General packet radio services
GSM	Global system for mobile communication
HCS	Hierarchical cell structure
HSCSD	High-speed circuit switched data
ILR	Interworking location register
IMT	International mobile telecommunication
IN	Intelligent network
IP	Internet protocol
ISDN	Integrated services digital network
ITU	International Telecommunication Union
MAP	Mobile application protocol
MSC	Mobile switching center
PCS	Personal communications services
PDC	Personal digital cellular
PSN	Packet-switched network
PSTN	Public switched telephone network
RACE	Research and technology development in advanced communications technologies in Europe (1987-1995)
RF	Radio frequency
RNC	Radio network controller
SMG	Specialized mobile group
SMS	Short message service
TTC	Telecommunications Technology Council
UMTS	Universal mobile telecommunications system
VHE	Virtual home environment
WCDMA	Wideband CDMA
WIN	Wireless intelligent network

- Support for high-speed packet data, including the following Internet applications:
  - browsing information;
  - subscribing to information (news, weather, traffic) via push-techniques—the information can even be location-dependent;
  - remote and wireless access to the Internet/intranets;
  - electronic commerce applications.
- Messaging services, such as multimedia e-mail (photo/video postcards).
- Real-time audio/video applications, such as videophone, interactive videoconferencing; audio/music; and specialized multimedia business applications—for example, telemedicine and remote security surveillance.

New portable/pocketable wireless terminals will be used to support these new multimedia applications (Figures 4 and 5).

Third-generation requirements

The main requirements that apply to third-generation systems are:

- support for high data rates—up to at least 144 kbit/s (384 kbit/s) in all radio environments and up to 2 Mbit/s in low-mobility and indoor environments;
- support for symmetrical and asymmetrical data transmission;
- support for packet-switched and circuit-switched services, such as Internet (IP) traffic and real-time video;
- support for good voice quality (comparable with wireline quality);
- support for greater capacity and improved spectrum efficiency compared with existing second-generation wireless systems;
- support for several simultaneous services to end-users and terminals—that is, for multimedia service capabilities;
- support for the seamless incorporation of second-generation cellular systems and support for coexistence and interconnection with mobile satellite services;
- support for roaming, including international roaming, between different IMT-2000 operators;
- support for scale-of-economy and an open global standard to meet mass-market needs.

Evolution into third-generation systems and standards

Setting third-generation wireless standards

In the late 1980s, the International Telecommunication Union (ITU) formed a study group with the aim of evaluating and specifying requirements for future wireless standards for the delivery of high-speed data and multimedia services. The third-generation standard is now called IMT-2000, where IMT stands for *International Mobile Telecommunication*.

In Europe, the European Telecommunications Standards Institute (ETSI) is currently deliberating on a standard for a third-

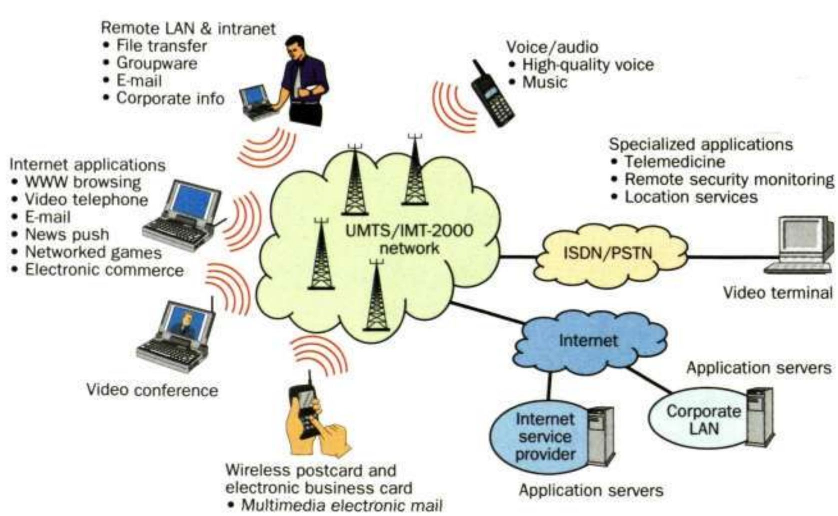


Figure 4
Examples of third-generation user applications.

generation pan-European system, called the *Universal Mobile Telecommunications System* (UMTS).

In Japan, the Association of Radio Industries and Broadcasting (ARIB) and the Telecommunications Technology Council (TTC) are also working on a third-generation wireless communication standard.

And in the USA, the American National Standards Institute (ANSI) is formalizing specifications for the evolution of D-AMPS/IS-136 and CDMA/IS-95. They will also propose UMTS in coordination with ETSI.

The major regional standardization bodies, ETSI, ARIB/TTC and ANSI, will submit their proposals to the ITU for possible harmonization and approval as IMT-2000 standards (Figure 6).

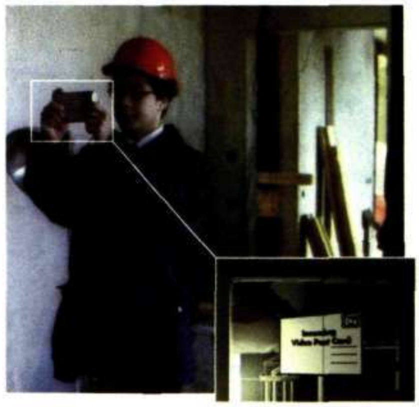


Figure 5
Third-generation-capable systems will offer an array of new applications over wireless networks. One example is video and digital postcards, which may apply to a range of consumer mass-market and specialized business segments.

Evolution of existing digital standards

The existing digital wireless standards continue to be developed, particularly as relates to value-added services, capacity, coverage costs and bandwidth. Three of the standards (GSM, D-AMPS/IS-136 and CDMA/IS-95) are expected to provide third-generation capabilities soon after the year 2000.

When third-generation systems become commercially available, more than 600 million subscribers are expected to use cellular services worldwide, which is a substantial customer base and represents considerable investment. Therefore, one of the most important requirements of the third-generation system is that it provide a seamless path of migration from present-day digital wireless networks (GSM, D-AMPS/IS-136, CDMA/IS-95 and PDC) and that it be capable of interworking with them.

All major providers of wireless network systems, services and terminals agree that future third-generation wireless systems should evolve from the core infrastructures contained in today's digital networks.

GSM enhancements

GSM will be enhanced to provide even better capacity, coverage, quality and data rates.

A series of developments is planned to enhance the functionality of GSM networks:

- One enhancement, called *customized application for mobile enhanced logic* (CAMEL), will give subscribers continued support of intelligent network (IN) services when they roam into other networks; for example, by creating a virtual home environment (VHE) for visiting subscribers. The first phase of CAMEL has already been implemented.
- The first enhancement for increasing data rates to reach commercial deployment will be high-speed circuit-switched data (HSCSD). Initially, this enhancement will support data rates up to 57.6 kbit/s using four 14.4 kbit/s time slots.
- General packet radio services (GPRS) is a packet-switched service that will allow full mobility and wide-area coverage with data transmission rates up to 115 kbit/s.
- Enhanced data rates for GSM evolution (EDGE) will use enhanced modulation and related techniques, further improving local mobility (typically in urban areas) with data rates up to 384 kbit/s. The existing GSM carrier bandwidth of 200 kHz will remain unchanged as will

Figure 6
Evolving toward global standards—IMT-2000.

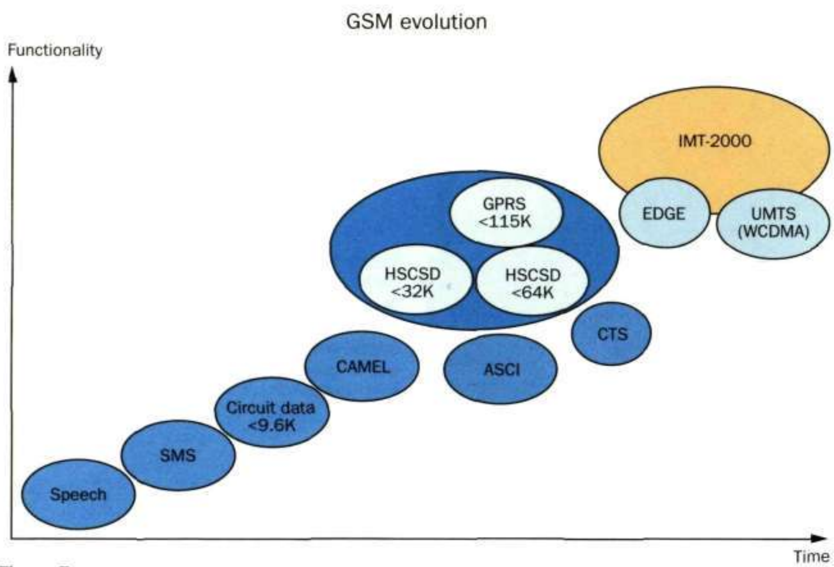


Figure 7
GSM evolution toward third-generation capabilities.

the complete time-division multiple access (TDMA) frame structure, logical channel structure, frequency plans and methods. Channels with EDGE functionality will be able to operate in either GSM/GPRS or EDGE modes. Coexistence of this kind in the same network will make it possible to introduce EDGE technology incrementally. Operators will be able to offer third-generation wireless services in any of today's GSM frequency bands: 900, 1800 and 1900 Mhz (Figure 7).

D-AMPS IS-136 enhancements

Two additional phases of enhancement are envisaged for the D-AMPS/IS-136 standard. In the first phase, denoted 136+, the bit rate of the 30 kHz radio carrier will be increased through the use of high-level modulation to achieve bit rates up to about 64 kbit/s.

The second phase, referred to as 136HS (high-speed), will involve a new air-interface specification based on EDGE technology. The ANSI IS-136 community has adopted an EDGE-based approach to providing high-speed data services. This solution will include steps toward harmonization with other EDGE standards-development organizations, such as ETSI/SMG and ANSI/T1P1, thereby fulfilling the IMT-2000 requirement for data rates up to 384 kbit/s (and indoor rates up to 2 Mbit/s with a proposed larger carrier).

In addition to increased data rates, D-AMPS technology will be enhanced to support better capacity, coverage, voice quality and value-added services (Figure 8).

Existing standards evolving to give third-generation capability

When fully evolved, the GSM and D-AMPS air-interface standards will be IMT-2000 systems using EDGE technology. As such, they will allow—with comparatively little new hardware and software upgrades—third-generation wireless services to be offered in any of the existing frequency bands used for GSM or AMPS/D-AMPS. The use

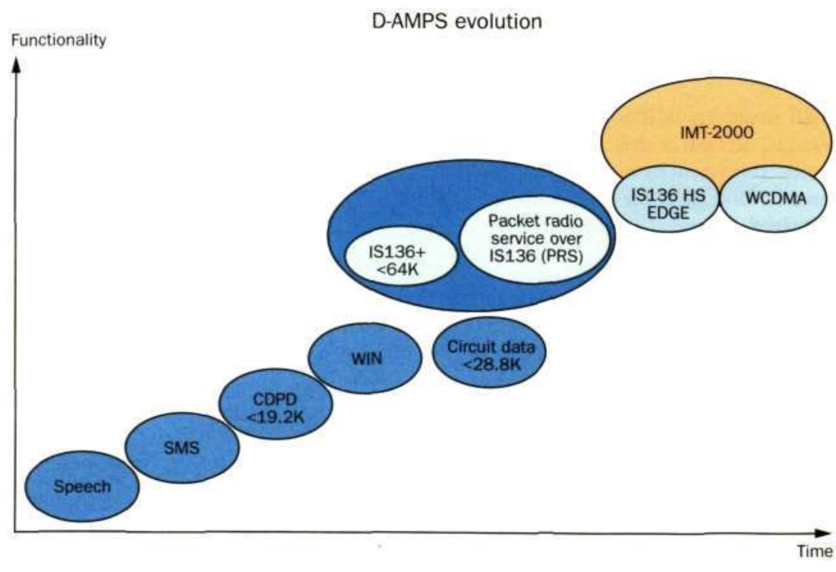


Figure 8
D-AMPS evolution toward third-generation capabilities.

of EDGE-based technology for GSM and D-AMPS standards simplifies the development of world terminals with global roaming capability and wireless multimedia applications.

In addition to the evolution of existing air-interface standards, a new, optimized third-generation radio access is being standardized, initially for the new 2 GHz spectrum band for IMT-2000 (Figure 9).

New, global WCDMA radio technology

ETSI has chosen WCDMA as the radio technology for UMTS in the paired band and TDMA/CDMA in the unpaired band. The paired band is used primarily by licensed public operators who provide wide-area communication services; the unpaired band may be used for unlicensed and licensed services that provide local or indoor communication. To provide the new WCDMA radio access, ETSI has decided to base the UMTS core network on the core switching network evolved from GSM. The focus of this article is on WCDMA for the paired band.

The Japanese standardization body, ARIB, has adopted this same WCDMA technology, initially using a 5 MHz carrier bandwidth. In addition, Japan and the world's largest mobile operator, NTT DoCoMo, have committed to implement ETSI's evolved GSM core network as the third generation core network.

Furthermore, the ITU has allocated a new 2 GHz frequency band to third-generation services (IMT-2000) being used in Europe and most of Asia (including Japan). However, WCDMA can also be deployed in existing refarmed frequency bands (initially, 2x5 MHz minimum).

Incorporation of second-generation systems

The new WCDMA access will coexist with existing and evolved GSM access and, with the help of dual-mode mobile terminals, will support full roaming and handover from one system to another. The use of dual-mode terminals in the introductory phases of WCDMA ensures that subscribers can roam and interwork with the rest of the GSM community from the very outset (Figure 10).

In Japan, the plan is to deploy IMT-2000 as a fully overlaid network on top of the PDC

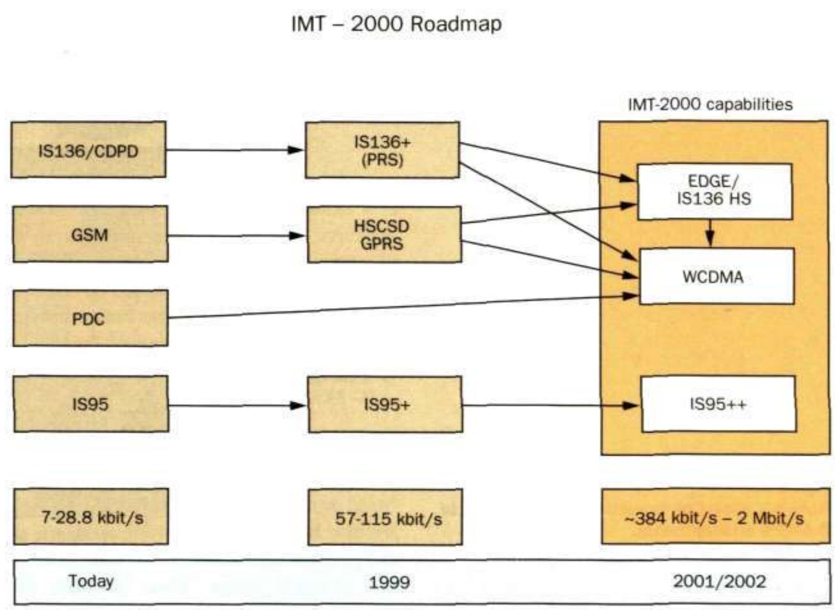
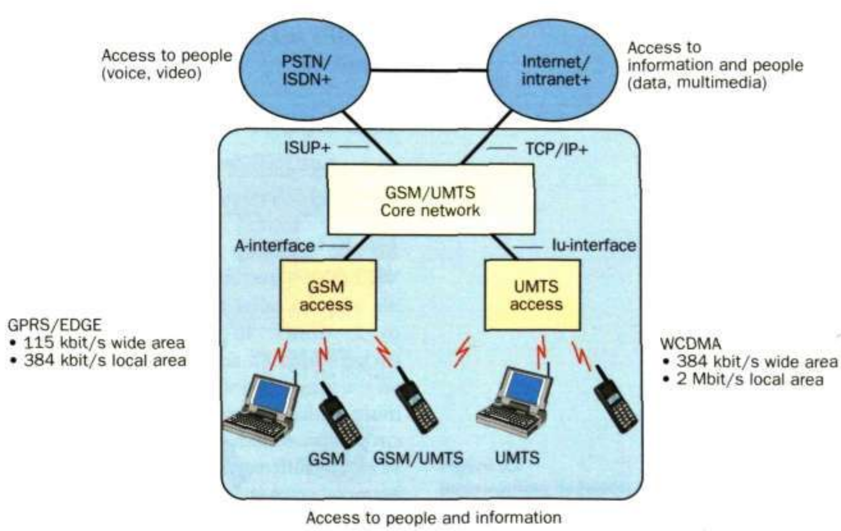


Figure 9
Evolution of digital standards.

Figure 10
Network evolution toward third generation in GSM environments.



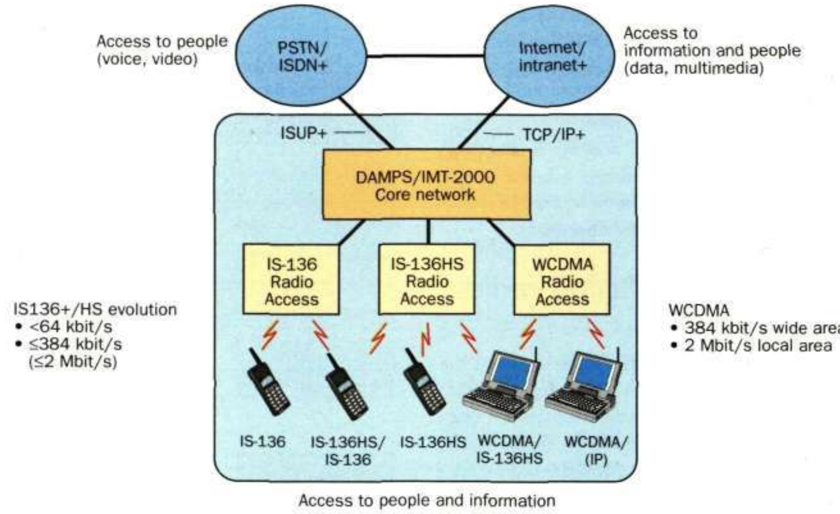


Figure 11
Network evolution toward third-generation in D-AMPS environments.

network, with interworking functions between the two networks and possibly PDC/IMT-2000 dual-mode terminals.

The D-AMPS/IS-136 community is also looking into opportunities to add the new, global WCDMA radio access, at least for Internet-based multimedia applications (Figure 11).

The GSM migration strategy with dual-mode terminals will be a generic migration strategy between third-generation WCDMA and second-generation systems.

A common product strategy—which will be based on the Internet protocol, EDGE, WCDMA, interworking units (ILR) and dual-mode terminals—will enable the D-AMPS and GSM standards to interwork (Figure 12).

The benefits of WCDMA

The wideband CDMA air-interface technology offers a range of features and benefits.

Service flexibility

WCDMA allows each 5 MHz carrier to handle mixed services ranging from 8 kbit/s up to 2 Mbit/s. In addition, circuit- and packet-switched services can be combined on the same channel, thereby allowing true multimedia service with multiple packet or circuit connections on a single terminal. Services with different quality requirements—for example, voice and packet data—can be supported with excellent capacity and coverage.

Spectrum efficiency

WCDMA makes very efficient use of available radio spectrum. Frequency planning is unnecessary, since *one-cell reuse* is applied. Other techniques, such as hierarchical cell structures (HCS), adaptive antenna arrays and coherent demodulation (bidirectional), can also increase network capacity. A two- or three-layer network can be deployed within the 2 x 15 MHz frequency allocated to operators, since each cell layer solely requires 2 x 5 MHz.

Capacity and coverage

WCDMA radio frequency (RF) transceivers can handle eight times more voice traffic than narrowband transceivers. Each RF carrier can handle approximately 80 simultaneous voice calls, or 50 simultaneous Internet-type data users per carrier. The capacity of WCDMA is approximately double that of narrowband CDMA in urban and suburban environments. The wider bandwidth and use of coherent demodulation and fast power control in the uplinks and downlinks yield a lower receiver threshold. The coherent demodulation and wider bandwidth also improve coverage.

Capacity can be improved further by adding support for hierarchical cell structures, adaptive antenna arrays and multi-user detection.

Hierarchical cell structures use a new hand-off method, called mobile-assisted interfrequency hand-off, between WCDMA carriers. Adaptive antenna arrays optimize the antenna pattern for each individual mobile terminal, improving spectrum efficiency and capacity. Multi-user detection decreases interference within a cell and improves capacity.

Improved voice capacity

Third-generation wireless access is also a very spectrum-efficient mechanism for voice traffic. For instance, operators with a 2 x 15 MHz spectrum allocation will be able to handle at least 192 voice calls per cell sector.

Multiple services per connection

WCDMA meets true third-generation requirements, allowing packet- and circuit-switched services with variable bandwidths to be mixed freely and delivered simultaneously—while meeting specific quality levels—to the same user. Each WCDMA terminal can access multiple services at the same time, including voice or a combination of data services, such as fax, e-mail and video.

Fast service access

To support instant access to multimedia services, a new random-access procedure has been developed that uses fast synchronization to handle 384 kbit/s packet-data services. The procedure requires only a few tenth milliseconds to set up connections between a mobile user and a base station.

Asynchronous radio access

Since WCDMA has its own internal system for synchronizing radio base stations, it is not dependent on external system synchronization—for example, IS-95 depends on the global positioning system (GPS) for synchronization. Dependencies of this kind for indoor radio base stations can make implementation difficult and expensive.

Economies of scale

The addition of WCDMA wireless access to a digital cellular network and interworking between the two systems permit the existing core network to be reused; what is more, in many cases, the current base station sites may also be used. Links between the WCDMA access network and the existing core network use ATM adaptation layer 2 (AAL2), which is the latest ATM minicell transmission protocol. This highly efficient way of handling data packets increases the capacity of a standard E1/T1 line to approximately 300 voice calls, compared with 30 using present-day networks, and saves up to 50% on the cost of transmission.

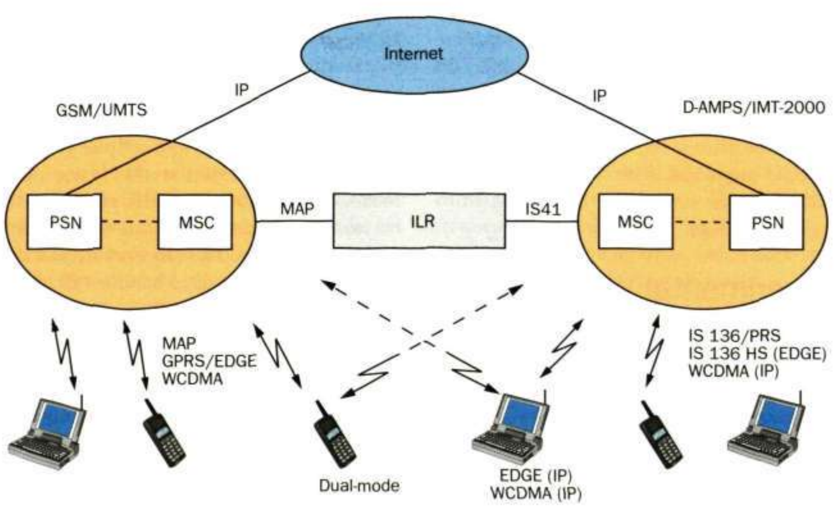


Figure 12
Interworking between GSM, D-AMPS and IMT-2000.

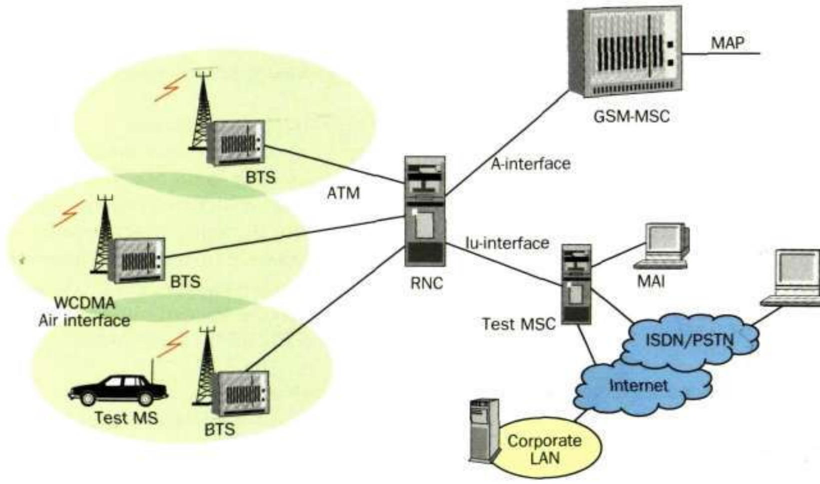


Figure 13
Overview of the experimental WCDMA systems.

Seamless access
Dual-mode terminals will provide seamless handover and roaming with mapping of services between GSM or D-AMPS and IMT-2000 networks. The choice of WCDMA—in mobile communication for UMTS/IMT-2000 in Europe, Japan, and for most GSM operators—offers a unique opportunity for creating a harmonized global standard with seamless global roaming for next-generation services.

Low-risk, mature technology
WCDMA is close to commercial deployment in Japan (2001), following WCDMA test-bed work carried out by NTT DoCoMo. The test systems are being supplied by several major European, US and Japanese manufacturers. In Europe, work on WCDMA began in 1989 as part of research and technology development in advanced communications technologies in Europe (RACE programs, 1987-1995). European WCDMA test systems are also expected to appear very soon.

Experimental WCDMA system

Ericsson has built an experimental WCDMA system as a facilitator for IMT-2000 standardization work (Figure 13). The system makes it possible to demonstrate and evaluate new third-generation services and technical solutions and offers an excellent opportunity for evaluating WCDMA characteristics.

The experimental system consists of a test mobile services switching center (test MSC), a radio network controller (RNC) and three base transceiver stations (BTS) with up to six sectors and two 5 MHz carriers each. A test mobile station is used for system testing and evaluation.

The radio network controller, which supports an A-interface connection to a mobile switching center in a GSM network, allows calls between GSM and WCDMA terminals.

The main task of the test mobile switching center, which has simplified call-control and mobility-management functionality, is to set up calls to, and release calls from, mobile stations. The node is based on Ericsson's new, generic ATM switch infrastructure, which handles packet-based traffic cost-effectively in transport and cellular systems.

Each base transceiver station is equipped to handle up to 300 voice calls, and the radio network controller supports a total of 400 mobile terminals. The test mobile switching center is equipped to handle 140 external connections.

Services in the experimental WCDMA system will be released in stages, starting with an 8 kbit/s voice service and 64 kbit/s circuit-switched data, followed by more advanced experimental services, such as 64 to 384 kbit/s circuit-switched data, high-quality voice codec, packet-switched data

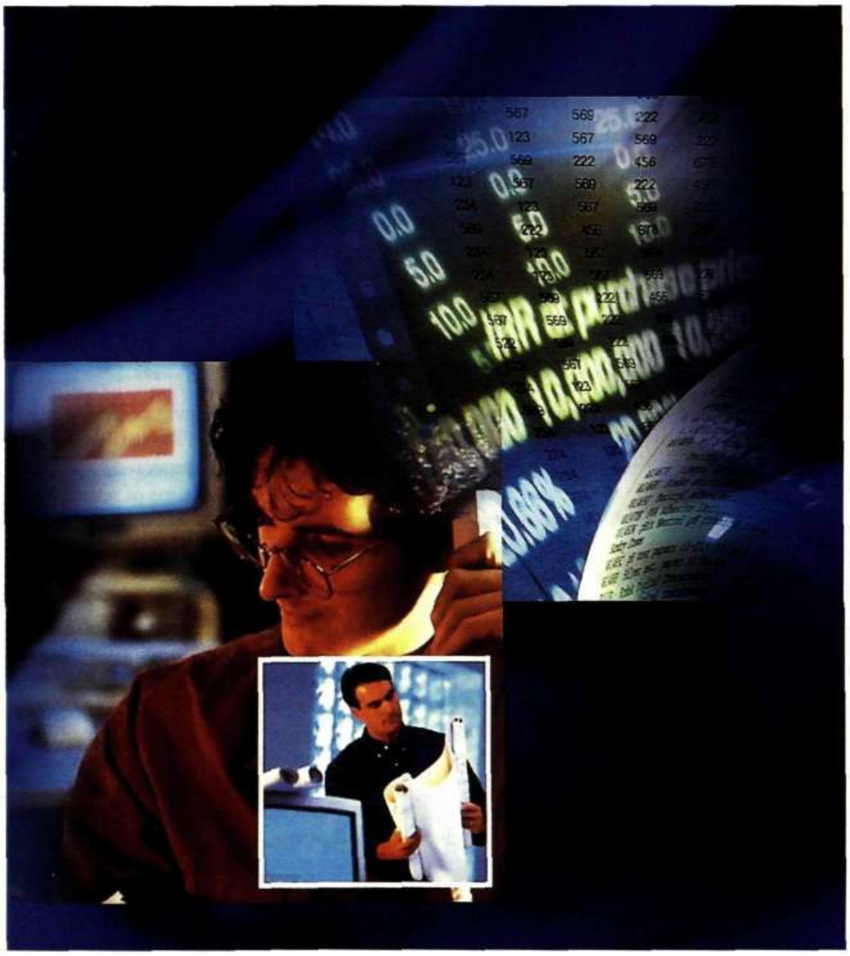


Figure 14
Futuristic world of multimedia.

up to 384 kbit/s and multimedia support in mobile terminals (Figure 14).

Conclusion

The world of wireless communication stands on the threshold of a golden era where true wireless multimedia services can be offered seamlessly on a global scale.

The choice of air-interface technology for connecting radio base stations and subscribers is a determining factor in realizing the full potential of the third-generation wireless vision. The choice must take into account the best interests of current digital network operators, end-users and the telecommunications industry as a whole.

Present-day GSM and D-AMPS networks will continue to be at the heart of wireless services as third-generation services are introduced. These digital wireless standards are rapidly evolving so that by the year 2000/2001 they too will be able to support third-generation services within existing frequency allocations.

Where third-generation, wide-area services are introduced, the core networks will have evolved from GSM or D-AMPS standards. In particular this applies to initial stages in the new 2 GHz bands, which use WCDMA as the new wide-area and wideband air interface. This approach gives wireless network operators a choice of migration paths toward third-generation, wideband multimedia services.

Wireless intelligent networks—The flexible future

Robert Foster

Wireless intelligent network techniques allow new communication services to be quickly developed and introduced across a wireless network without requiring major upgrades.

In addition to creating revenue-earning opportunities for wireless network operators, the new services give end-users a high level of personal control over their communication services.

The author describes some of the WIN-based services becoming available for wireless networks based on the D-AMPS/IS-136 standard.

Although the promise of intelligent network (IN) techniques has been widely discussed in recent years, IN-based services are only now becoming broadly available to end-users.

An intelligent network is not in itself a product or technology, but describes a set of extra capabilities that can be added to a fixed or mobile telecommunications network, thereby enabling network operators to tailor their communication services to suit the precise needs of end-users. Operators can also generate extra streams of revenue, by creating and introducing new services quickly.

From a network perspective, the intelligent network concept depends on separating the intelligence within a telecommunications service from the physical network infrastructure, particularly from switching elements. Separation of this kind gives operators enormous opportunity and flexibility in providing end-user services, since the services cease to be determined by characteristics of the physical network.

In the past, the introduction of new, end-user services in fixed telecom networks meant making major changes at every local exchange in the network. As a result, new services were costly and slow to implement. In the IN environment, however, the network infrastructure is simply a delivery channel that gives all end-users access to services. Thus, services can be developed quickly and introduced wherever needed—either in specific areas or across the entire network.

IN-based services are expected to give network operators new streams of revenue and to give end-users a range of attractive communication options. As end-users seek to control communication costs and make full use of the potential benefits of the *one phone, any place, any time* concept, the new service options might actually have greater appeal in a wireless network environment than in a fixed network.

Most articles on intelligent networks approach the subject from a network point of view, examining how the hardware and software functions needed for developing, deploying and managing IN-based services are arranged. This article adopts a different approach, looking at new wireless intelligent network-based (WIN) services that will allow end-users to harness the power of their mobile phones (and in the future, more sophisticated wireless terminals) in ways that suit their business and personal lifestyles. Some supporting information is presented on network infrastructure and service-management perspectives, and several possibilities for innovative services are also discussed.

New service opportunities

Because intelligent networks are about end-user services, it makes sense to approach the subject by first looking at new service opportunities.

Some IN-based services, such as freephone and credit-card calling, are well-known and understood. Moreover, it is becoming clear that these are only the first of what will become a broad range of service possibilities. Many new services will be developed by combining basic IN services in innovative ways.

To appreciate the potential, it is interest-

Box A, Abbreviations

ACD	Automatic call distribution
AIN	Originating services, subscriber category
BIN	Terminating services, subscriber category
BNT	Bulk number translation
CIN	Transferring services, subscriber category
CDPD	Cellular digital packet data
CPCC	Calling-party-controlled completion
D-AMPS	Digital advanced mobile phone service
DP	Dual profile
EUC	End-user control
FCF	Flexible call forwarding
HLR	Home location register
IN	Intelligent network
IS-136	Base station to the mobile terminal air interface
MSC	Mobile switching center
OCA	Outgoing call allowance
OCR	Outgoing call restriction
OSF	Open Software Foundation
PBX	Private branch exchange
PCS	Personal communications services
PNP	Private numbering plan
SCA	Selective call allowance
SCF	Selective call forwarding
SCP	Service control point
SCR	Selective call rejection
SES	Service provision subsystem
SFW	Service framework
SMA	Service management application
SMAS	Service management application system
TFC	Toll-free calling
TMOS	Telecommunications management and operations support
WIN	Wireless intelligent network
WVPN	Wireless virtual private network

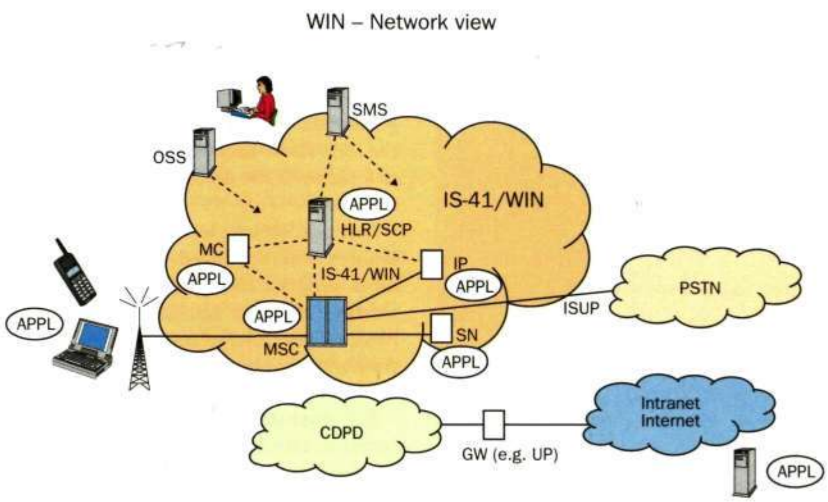


Figure 1
WIN network view

APPL	**Application (or part of an application)**
GW	**Gateway**
HLR/SCP	**Home location register/service control point**
IP	**Intelligent peripheral**
IS-41	**Interim standard - 41**
MC	**Message center**
MSC	**Mobile switching center**
OSS	**Operations support system**
SMS	**Service management system**
SN	**Service node**
UP	**Unwired planet**
WIN	**Wireless intelligent network**

ing to look at two examples of services that WIN technology permits. The first is the application of IN techniques for controlling the cost exposure of mobile phone subscribers as they roam into wireless networks outside their home region or country.

As mobile subscribers travel abroad, calls made to their mobile phones from the home region are automatically forwarded to the network in which they are currently located. Obviously, this is one of the main advantages of wireless communication and automatic roaming.

The potential drawback, however, is that subscribers must pay for the leg of the call from their home network to the destination network. As a cost-control mechanism, users may want to exercise control over the calls, forwarding some and redirecting the rest to a voice mailbox or recorded message.

A typical scenario would be for subscribers to set up an incoming call-control profile so that only calls from their office or home, or from specific business contacts, are forwarded. Of course, the profile can be changed at any time, even as subscribers travel.

The second example illustrates how WIN technology can create completely new business opportunities—not just for wireless network operators. Thanks to a new communication service, which will be especially attractive to business people, hotels can greatly enhance the quality of service they offer their guests.

In business-class hotels, guests are accustomed to having access to voice-mail functions via their room phone. While helpful, this solution is far from perfect. For example, hotel guests returning to their rooms at the end of the day, often have little opportunity to respond to messages on their voicemail, particularly when the calling party is several time zones away.

But if, instead of voice mail, hotel guests were offered the use of wireless phones that behaved like wireless extensions to the hotel, then any standard D-AMPS/IS-136 mobile phone could be used. And thanks to the use of a common wireless standard for in-building and outdoor access, the same phone could also be used by guests who leave the vicinity of the hotel—say, to attend a business meeting or to dine downtown.

With the addition of WIN technology, the hotelier has all the functions needed to offer this service to guests—including all necessary billing functions. The service is activated when guests check in and deactivated when they check out. The hotelier could do this by making a profile change from the mobile phone itself.

These and many other new services based on WIN technology will make wireless communication increasingly flexible for users and can be created by combining various basic IN functions and services.

WIN service basics

The starting point for IN services in wireless networks is a range of basic subscriber services that can be activated at different stages in the progress of a call.

Box B
Originating services

The following originating services are currently available:

End-user control (EUC)
This service allows the subscriber to select one of five different profiles for managing calls to be transferred and for restricting outgoing calls.

Wireless virtual private network (WVPN)
This application, which applies specifically to corporate users, gives users a short-code dialing plan that fits the company PBX numbering scheme and gives every member of the user group access to the plan. The bulk number translation function (see the next item) is also part of this plan.

Bulk number translation (BNT)
This service allows a subscriber to define an abbreviated dialing code for a range of numbers, such as extensions to a business PBX. BNT is a call-translating service that reduces the number of required keystrokes by shortening the common part of the number to a two-digit code.

Private numbering plan (PNP)
This service, which is available to individual users, allows users to set up abbreviated dialing codes for up to 50 frequently called numbers. Typically, a four-digit code suffices. The private number plan is potentially useful for groups of business people, such as consultants or persons who work at home or do not share an office with others; it might also be used as the basis of a *family-or-friends* type of service for private individuals.

Outgoing call allowance (OCA)
According to this service, outgoing calls are only allowed if the called number is on a list of authorized numbers set up by the subscriber. Up to 35 numbers or number series, each with up to 20 digits, can be included in a profile. As an extension to the service, up to five different profiles can be established, allowing users, via end-user control, to set up different call-screening lists for home, the office, and other situations.

Outgoing call restriction (OCR)
This service allows users to set up a list of B-numbers to which calls should be blocked. If the phone is used to dial one of the blocked numbers, the call is routed to an announcement machine that informs the caller that the call could not be routed as dialed. All other calls are routed as dialed. Up to 35 numbers or number series can be selected for blocking. As with outgoing call allowance, five different profiles can be set up.

- Originating services, also known as AIN services (Box B), are used by the person making the call.
- Terminating services, known as BIN services (Box C), are used by the recipient of the call.
- Transferring services, or CIN services (Box D), are used under call-transfer conditions.

Box C
Terminating services

The following terminating services, which currently consist of call-screening services, are available:

Selective call acceptance (SCA)
Incoming calls are accepted only if they are on the SCA screening list of authorized numbers. Up to 35 numbers or number series can be authorized, each with up to 20 digits.

Selective call rejection (SCR)
Incoming calls from specific telephone numbers are rejected; all other calls are allowed. Up to 35 numbers or number series, of up to 20 digits, can be specified on the screening list.

Developing made-to-measure communication solutions

The real interest in WIN-based services lies not so much in the basic services mentioned above, but in the services that result from combining or marketing them in a way that helps subscribers to recognize a new potential benefit.

A main market sector of one of the chief targets is corporate accounts, where businesses can perceive the benefits from new communication services, such as improved productivity and higher levels of customer service. Opportunities exist for developing communication packages that are relevant to the needs of different staff levels in an organization, from the board of directors to the shop floor. Each service package can be tailored to fit the needs of users at each specific level.

In a similar way, service packages can be tailored to suit the needs of residential end-users, ranging from high net-worth individuals for whom call costs are not a big issue, to low-volume users who mostly use their wireless phones for incoming calls and security.

For instance, the combination of bulk number translation (BNT) and private numbering plan (PNP) services creates a communication environment that resembles a wireless office or virtual PBX for a closed group of users. The setting up of a user group with short-code dialing means that the group can be integrated into an existing PBX numbering scheme. Users of a fixed-extension would then be given access to users of the wireless extension by dialing the appropriate four-digit "extension." This functionality will work regardless of whether users of the mobile extension are located within the boundaries of their home network or in a roaming network.

Many companies still prefer radio pagers, because the cost of calls is fixed. Nonetheless, the outgoing call allowance (OCA) service in a D-AMPS network allows staff members to be equipped with wireless phones instead of pagers and keeps calling costs under control. For example, the wireless phone might be set up to receive any incoming call, but restricted solely to allow outgoing calls to a supervisor.

The selective call acceptance (SCA) service can be used to set up a group of wireless phones in a wireless automatic call distribution (ACD) system. Calls can be directed to staff members according to various parameters. One option is to direct calls according to geographical location, so that they are handled by persons with relevant local knowledge.

In another application of this capability, a company might have separate help desks for its corporate and private customers: the A-numbers of the corporate customers could be set up in the authorized list so that calls from these numbers are always directed to the corporate help desk. The staff taking these incoming calls do not have to be tied to their desks while waiting for calls. In fact, they do not even have to be in a defined location. They could be anywhere within reach of the wireless network.

A hotel environment provides greater potential for wireless phones than the guest service application mentioned above. The introduction of a private numbering plan, for example, could allow D-AMPS wireless phones to replace the two-way radios currently used in many hotels. Staff-to-staff calls would require only a four-digit number to be keyed into the phones. As a cost-control mechanism, OCA and outgoing call restriction (OCR) could be applied to restrict the destinations to which staff members can call using the phones.

Box D
Transferring services

The following transferring services are currently available:

Selective call forwarding (SCF)
This service allows a subscriber to have incoming calls forwarded to one of three different destinations, depending on which screening list the calling number appears. A maximum of five profiles can be defined. The subscriber can select them at different times to suit different situations; for example, when he or she is in a meeting or at home. The end-user control service could be used for activating the service or for changing the selected profile. If the calling number does not appear on a screening list, it is routed to a default destination; for example, to voice mail or a secretary.

Flexible call forwarding (FCF)
This service allows a subscriber to set up a forwarding schedule (time-of-day/day-of-week) in order to forward incoming calls to various destinations. This is a useful service for staff who work rotating shifts; for example, doctors and nurses.

Toll-free calling (TFC)
This service is the same as FCF except that the B-party (the receiving party) is charged for calls.

Field trials

Ericsson has been working closely with Telecom New Zealand to test-market personal communications services (PCS) based on WIN technology. Tests have included several innovative services built on generic IN services.

For instance, calling-party-controlled completion (CPCC) gives callers a range of options when the wireless phone they are calling cannot answer. The options are to route a call

- to voice-mail;
- to either of two forwarding numbers (typically a pager or an office number);
- to an operator.

Subscribers with CPCC can modify current settings to activate or deactivate the service, modify the forwarding numbers, and specify call-routing and call-flow features from their wireless phones or from any other telephone.

The dual-profile (DP) service allows subscribers to set up a business profile and a personal profile for their wireless phones. Typically, this would be arranged so that during normal business hours, when the business profile applies, all calls are charged to the business number. Calls made outside this period are charged to the personal account. For each profile, the subscriber must specify screening lists for the telephone numbers and area codes that can or cannot be called.

The parallel alerting service allows a wireless subscriber to specify several telephone numbers that ring simultaneously when the wireless phone is called. The call is completed to the first terminal that answers. In the event that no phone is answered, the incoming call can be directed to a forwarding destination (such as to an answering machine). Typically, the subscriber's home or office phone would be specified as the forwarding number.

The WIN architecture

In Ericsson's CMS88 system for D-AMPS/IS-136 wireless networks, the WIN system service logic is installed in a service control point (SCP), co-located with a home location register (HLR) and connected to a mobile switching center (MSC). The service control point makes all decisions regarding call screening, short-code dialing and end-user control before it passes the call to the mobile switching center, where it is routed.

The administration platform for the MSC is based on a Sun server with a terminal that

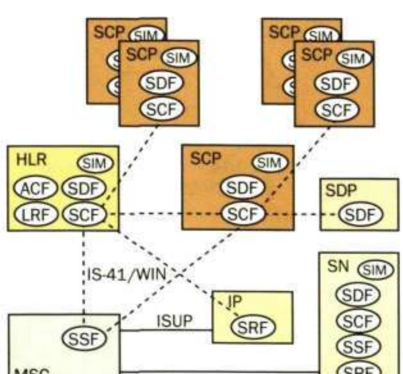
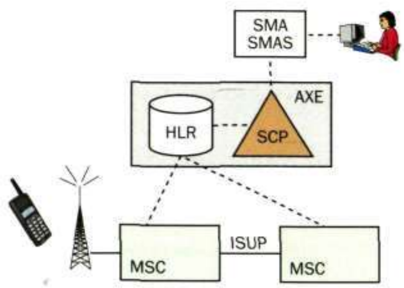


Figure 2
WIN architecture

HLR	**Home location register**
IP	**Intelligent peripheral**
MSC	**Mobile switching center**
SCF	**Service control function**
SCP	**Service control point**
SDF	**Service data function**
SDP	**Service data point**
SIM	**Service interaction management**
SN	**Service node**
SRF	**Special resource function**
SSF	**Service switching function**
WIN	**Wireless intelligent network**

Figure 3
C-package (version 3) integrated HLR/SCP configuration.



connects to the service control point. Systems may be interconnected by use of the IS-41 standard.

The WIN architecture is made up of six platform elements:

- The home location register/service control point (HLR/SCP) is a co-located network node that runs on the AXE platform. The main task of the home location register is to track the location and service profiles of wireless subscribers and deliver call information to interrogating switches for routing. The service control point executes the WIN service scripts.
- Telecommunications management and operations support (TMOS) is used by the service management application system (see the next item) to store all WIN service information and to communicate with the mobile switching center.
- The service management application system (SMAS) is a TMOS application that enables the operator to install, administer and maintain WIN services in the network. SMAS provides a graphical user interface for service maintenance.
- The service management application (SMA) is a graphical user interface that easily allows operators to provision and manage WIN subscribers and service subscriptions. It has a client/server architecture that supports multiple platforms, including OSF/Motif on Sun workstations and Microsoft Windows on personal computers.
- The service provision subsystem (SES) contains the intelligence required to execute the logic of IN services.
- The service framework (SFW) handles fault situations and determines how end-users connect to the WIN services. It is installed in the SES subsystem and is common to all services.

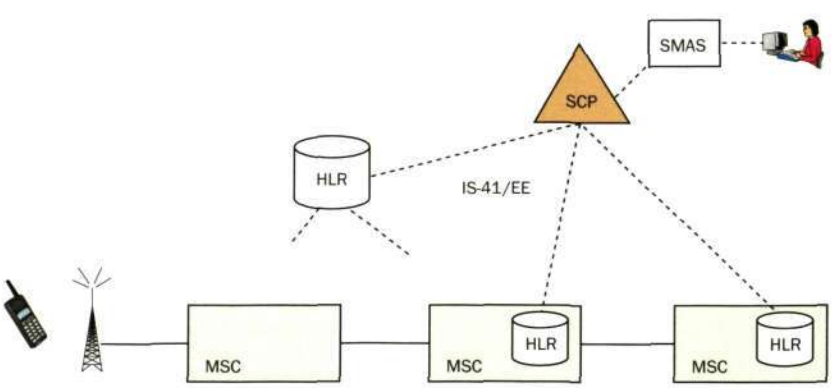
Conclusion

Once solely an issue of technical network infrastructure, the subject of intelligent networks has crossed over into the marketing domain.

Wireless IN techniques enable a broad scope of service possibilities that are attracting the attention of network operators. WIN technology creates opportunities for building market share and reducing churn by marketing innovative services tailored to fit the individual needs of different types of people and groups in business and residential applications.

This article has hinted at a few service possibilities, providing a foretaste of the services that will emerge as WIN technology becomes more widely deployed in wireless networks.

Figure 4
C-package (version 3) stand-alone SCP configuration.



Mobile crosstalk control—Enhancing speech quality in digital cellular networks

Anders Eriksson, Maria Eriksson, Tõnu Trump and Teresa Vallon Hulth

Although today's mobile phones satisfy recommendations for echo attenuation, under certain conditions echo that originates from the acoustic crosstalk in mobile handsets can still be noticed by users. Acoustic crosstalk echo and network echo demand different approaches. Hence, Ericsson has developed mobile crosstalk control, a special algorithm that eliminates acoustic crosstalk echo and enhances speech quality in digital cellular networks.

The authors briefly describe echo in the PSTN and how network echo cancellers work. They explain the fundamental differences between network echo and echo that originates from mobile digital handsets, the particular requirements for eliminating acoustic crosstalk echo in digital cellular networks, and the methods that mobile crosstalk control uses to eliminate echo. The authors also discuss the challenge of evaluating the speech quality that results from a specific algorithm and the variety of situations that an algorithm must be able to handle.

Echo in digital cellular networks

Cellular operators have identified speech quality as being a significant competitive factor in today's market, which has spurred developments aimed at removing disturbances from the voice channel in cellular systems.

A major cause of disturbance in telephony systems is echo, which occurs when part of a speaking party's voice signal energy is reflected back to him or her (Figure 1). Echo with a substantial delay (physical or processing delay in the transmission path) is irritating in telephone conversations. A typical source of echo is the impedance mismatch in the 4-wire to 2-wire conversion in PSTN subscriber interfaces. Today, it is common for international and mobile switching centers to employ echo cancellers for controlling echo generated in the local loop on the PSTN end of the connection.

Echo can also occur in cellular networks as the result of acoustic crosstalk inside mobile handsets due to the acoustic coupling between the microphone and loudspeaker of the digital handset. Echo of this kind is best controlled within the handset, as is recognized in several international recommendations. For example, the GSM specification requires that echo attenuation (measured in the switching system as the path loss from the input to the speech coder to the output from the speech decoder) should measure at least 46 dB.[1] Because transmission in digital cellular systems is "4-wire" throughout, mobile telephones are usually not regarded as a potential source of echo in the system.

Although handsets satisfy requirements for echo attenuation, under certain conditions users can still notice echo that originates from acoustic crosstalk. This is mainly due to two factors:

- The test specification does not take into account all possible variations in the position of the handset during a normal conversation.
- The line levels within the telephone system may deviate from nominal levels.

Echo from acoustic crosstalk can be annoying to users. If the echo can be eliminated, the overall speech quality of the system will improve. However, because acoustic-crosstalk echo differs greatly from conven-

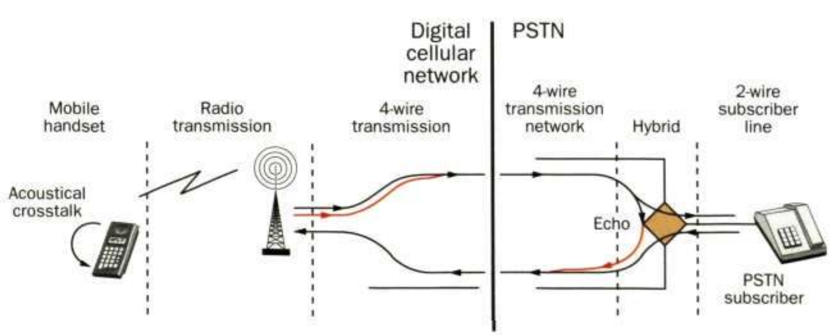


Figure 1
Sources of echo that affect digital cellular networks.

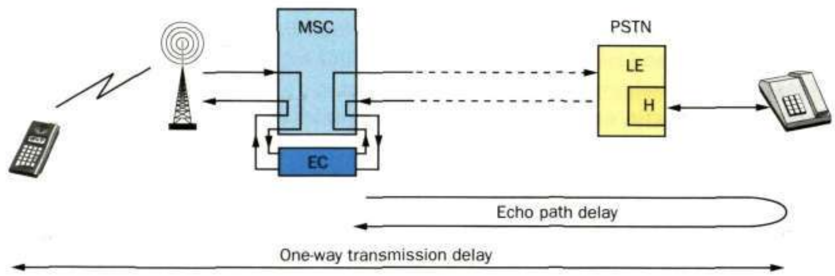


Figure 2
Location of network echo cancellers in digital cellular networks.
MSC **Mobile services switching center**
LE ***Local exchange***
H **Hybrid**

Box A
Abbreviations and definitions

CNG
Comfort noise generator.
Comfort noise
A signal—whose characteristics are similar to the background sound—that is *added to the output of an echo canceller* in order to mitigate the undesired effects of the non-linear processor.
Conversation test
Test in which test persons carry on a conversation, simultaneously evaluating the speech quality of the conversation.
D-AMPS
Digital advanced mobile phone service.
Downlink
Transmission from a base station to a mobile handset in a cellular network.
DSP
Digital signal processor.
DTX
Discontinuous transmission. To save power, the transmitter turns off when no one is speaking into the handset. This function is called DTX in GSM and D-AMPS and voice-operated transmitter (VOX) in PDC.
EC
Echo canceller.
ECP 303
Ericsson echo canceller in pool model 303.
ECP 323
Ericsson echo canceller in pool model 323 (model 303 featuring MCC).
ECP 404
Ericsson echo canceller in pool model 404.
ECP 424
Ericsson echo canceller in pool model 424 (model 404 featuring MCC).
ERL
Echo return loss. The attenuation of the signal as it passes the echo path.
ERLE
Echo return loss enhancement. Attenuation obtained by linear filtering techniques in an echo canceller.
FIR
Finite impulse response.
GSM
Global system for mobile communication.
Handover
The process of a mobile phone changing cells while a call is in process. Handover is the term used in GSM. In D-AMPS and PDC, the corresponding term is handoff.
MCC
Mobile crosstalk control. An algorithm developed by Ericsson for deployment in the switching system. The algorithm solves the problem of acoustic crosstalk in hand-held mobile telephones.
NLP
Non-linear processor. A function in an echo canceller which further attenuates echo; for example, by completely or partially blocking the transmission signal.
PCM
Pulse code modulation.
PDC
Personal digital cellular.
PSTN
Public switched telephony network.
Uplink
Transmission from a mobile handset to a base station in a cellular network.

tional network echo, cancelling it requires different solutions.

Overview of network echo cancellers

Today, international and mobile switching centers employ echo cancellers to control echo generated in the PSTN local loop. The means of cancelling this type of echo have been the subject of extensive study.[2]

Ordinarily, the PSTN echo path can be accurately modeled using a linear finite-impulse-response (FIR) filter with coefficients that are constant or that vary slowly over time. The echo canceller is usually positioned in the system so that the duration of the echo path is less than 64 ms (Figure 2). An adaptive FIR filter with up to 512 coefficients is used for modeling the echo path. The echo path might change suddenly if the PSTN subscriber changes telephone sets or if a third party or additional equipment is connected to the call.

The attenuation of echo, which is measured in decibels, is called echo return loss (ERL). Echo return loss from the PSTN, which depends on the hybrid used in the network, can vary significantly depending on the local loop. According to K. Shenoi[3], the ERL can generally be regarded as a random variable picked from a Gaussian distribution. For the US network, the distribution has a mean value of 13.6 dB and standard deviation of 2.8 dB in a segregated loop-balancing scheme. Recordings taken from the Swedish PSTN show a similar distribution but a higher mean value (Figure 3).

The adaptive FIR filter is used to model the impulse response of the echo path (Figure 4). The filter yields echo return loss enhancement (ERLE), which reduces echo by about 20 to 40 dB. The reduction is limited by non-linearities; that is, the part of the echo that cannot be modeled using a linear filter. One source of non-linearities is the quantization of signals in the analog-to-digital converter. Other sources are possible non-linear effects (such as saturation) in hybrid or telephone sets. Figure 5 shows the optimum ERLE achievable using linear filtering techniques for different echo paths in the Swedish PSTN.

The adaptive algorithm used in the FIR filter can accurately estimate the coefficients, provided the signal from the PSTN consists mainly of echo. However, if speech or background sound from the PSTN dominates the echo—that is, if "double talk" exists—the algorithm returns a poor estimate of the echo path. What is more, the power of voice signals varies greatly from one moment to the next, meaning that the control part of the echo canceller must determine at each instant whether or not conditions are favorable for filter adaptation.

The echo that remains after linear processing, called residual echo, is often still audible. A non-linear processor (NLP) is used to remove it. When residual echo is expected, the NLP blocks the signal, either in part or completely. However, because the level of residual echo is not known in advance, the NLP must be adaptive, taking into account all possible variations in echo attenuation (the sum of echo return loss and echo return-loss enhancement). Consequently, the design of the NLP function is critical to user perceptions of the performance of echo cancellers. A well-designed NLP, which activates only if needed, should preserve speech and background sound from the PSTN end as faithfully as possible.

Finally, an echo canceller includes a comfort-noise generator (CNG) that generates a signal whose characteristics are similar to the background sound on the PSTN. When the NLP is active, the comfort-noise generator adds a comfort-noise signal to the output, which reduces any undesirable effects of the background sound modulation perceived by users.

Digital handset echo vs. network echo

From the viewpoint of the switching system, there are many characteristics that distinguish the echo that originates from the acoustic crosstalk in a mobile digital handset from network echo. Only minor signal

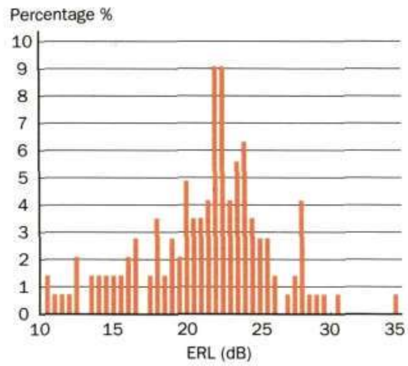


Figure 3
Histogram of echo return loss values measured for hybrids in the Swedish PSTN.

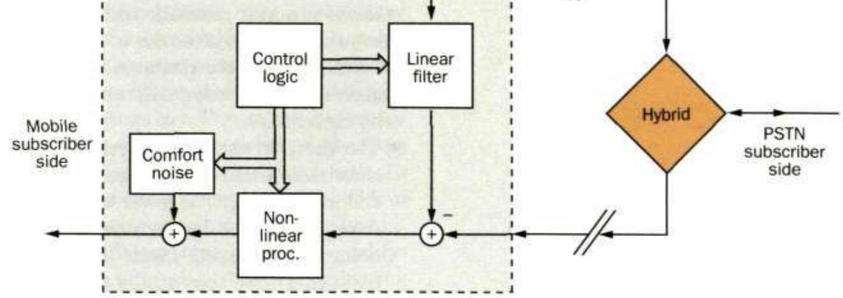


Figure 4
Echo canceller principles. A replica of the echo is obtained via a linear filter and subtracted from the PSTN subscriber signal. The residual echo signal is further suppressed using a non-linear processor.

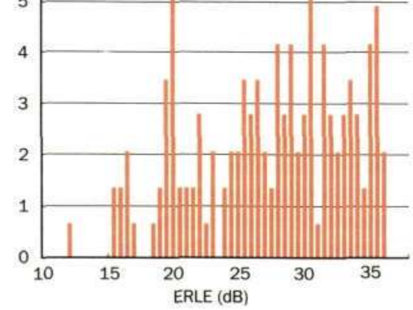


Figure 5
Histogram of achievable echo return loss enhancement (ERLE) for PSTN hybrids as measured in the Swedish PSTN.

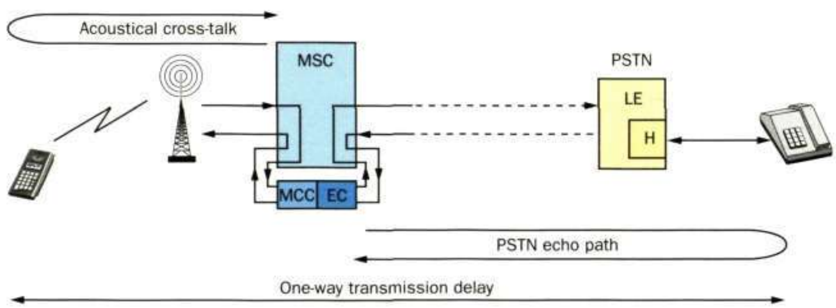


Figure 6
Differences in the echo paths of acoustic crosstalk and PSTN echo.
MSC Mobile services switching center
LE Local exchange
H Hybrid

disturbances occur in the echo path of the network echo canceller (Figure 6). However, the echo path of the echo canceller that controls acoustic crosstalk includes radio transmission. This gives rise to fundamental differences in the characteristics of the mobile echo path, compared with network echo cancellation.

- The delay in the handset echo is long, since radio transmission requires coding and interleaving. The lower limit of this delay is specified by the delays in each necessary processing block in the echo path. In a GSM system, the delay is approximately 180 ms. The actual echo delay for a call can vary, depending on the handset, system hardware, extra signal processing equipment, and the routing of the call.
- The characteristics of the mobile echo path are apt to vary rapidly and frequently due to changes in the position of the handset. The characteristics may also be affected by bit errors in the radio transmission, handover, or discontinuous transmission—in order to save power, the mobile unit might interrupt transmission; this function is called discontinuous transmission (DTX)—in this case, on the uplink. Discontinuous transmission in the uplink causes large variations in the characteristics of the echo path. The DTX function is activated for the uplink when only the downlink carries speech signals. Thus the handset produces no echo. Spurious echo bursts might be transmitted anyway, although the duration of the bursts is generally too short for them to be modeled successfully. When both parties talk simultaneously, the DTX function is not activated, in which case echo might be noticed. Owing to double talk, this echo cannot be modeled successfully by an adaptive algorithm.
- Non-linear effects are introduced by speech coding and bit errors in the radio transmission. For typical communication, speech coding and radio transmission reduce the ERLE achievable by a linear filter to less than 10 dB. The achieved ERLE, which to a large extent depends on the input signal, can vary significantly over a short period. This contrasts with the much more consistent ERLE achievable in a network echo canceller, which is limited to 38 dB because of PCM coding in the network echo path.

These problems make it significantly more complicated for the switching system to control acoustic crosstalk echo from a mobile handset than to control network echo.

At the same time, some other factors simplify the design of the acoustic crosstalk algorithm. First, the length of the acoustic crosstalk echo path (the dispersion), which is usually less than 5 to 10 ms, is relatively short compared with that of network echo. Second, the overall level of echo from the mobile handset is significantly lower than that of network echo, thanks to the design of the handsets. In most cases, the level of echo from the mobile handset is comparable to or lower than the level of residual echo after linear processing in a network echo canceller.

Taken together, the factors discussed above clearly show that the linear echo-path model used for designing a network echo canceller is not adequate for treating the problem of echo from digital handsets. Consequently, an ordinary network echo canceller will not produce good results for the latter type of echo, which demands a different algorithm.

Basics of the MCC algorithm

The basic requirement for any device used in the switching system for cancelling echo that originates from acoustic crosstalk in mobile terminals is that the device must never introduce any artifacts in calls when audible acoustic crosstalk is not present. Ericsson has developed an algorithm that can successfully operate under the conditions described in the previous section and that also meets this basic requirement. The

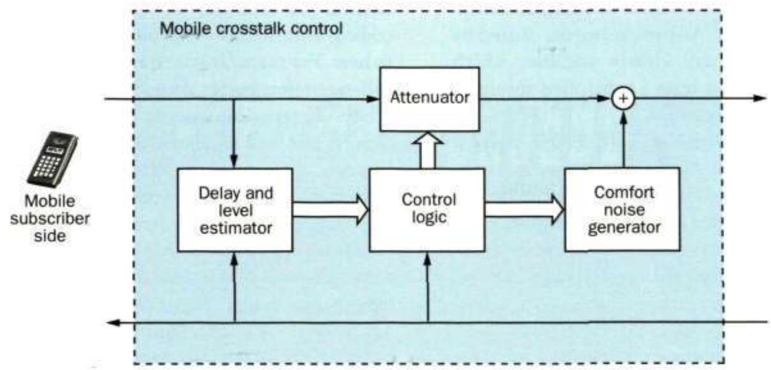


Figure 7
MCC principles. The delay and level of the echo is estimated. The estimates are then used for controlling an attenuator in the transmission path.

new function is called mobile crosstalk control (MCC).

The MCC algorithm (Figure 7) detects the presence of echo in the uplink signal and determines its delay and power. In order for the echo canceller to react properly to changes caused by hand-over, handset positioning, or the amplification setting on the handset, the delay and power of the echo must be estimated continuously during the call.

If echo is the dominant component of the uplink signal, and is regarded as annoying to the PSTN subscriber, the uplink signal is attenuated—but only for short periods, determined by an estimate of echo power. This ensures that the algorithm preserves the duplex quality of the mobile connection.

To prevent line noise or background sound from the digital handset from being modulated, carefully designed comfort noise is added to the uplink whenever the attenuator is activated.

Demands put on mobile crosstalk control

The task of mobile crosstalk control is exacting:

- It must cancel the acoustic crosstalk echo from the cellular phone.
- It must never degrade speech quality if echo is not present.
- It must never introduce clipping into or distort the incoming speech signal.

Evaluating the speech quality produced by network echo cancellers is difficult; evaluating the speech quality produced by an algorithm for cancelling echo from a mobile unit is even more difficult. Different methods are used for making subjective evaluations. Today, no methods for making objective measurements exist. Traditional methods of subjectively evaluating speech quality on transmission channels, such as methods used for evaluating speech coders, are not appropriate. Due to the duplex nature of the MCC algorithm, evaluations must involve simultaneous speech from the mobile handset end and the PSTN end, as is done when evaluating network echo cancellers. Different types of speech-quality-impairment must be defined for evaluating the echo-cancellation algorithm. At present, four classes have been defined:

- Echo—residual echo not handled by the algorithm.
- Clipping—loss of speech and background sound from the mobile handset end. In addition to attenuating the echo, the algorithm may unintentionally attenuate the speech and background sound from the mobile subscriber end.
- Distortion—unintentional modification (other than clipping) of speech from the mobile handset end.
- Faulty insertion of comfort noise—the insertion of comfort noise that is perceptually different from actual background sound.

The speech quality produced by the algorithm for cancelling echo from the mobile handset end must be evaluated in listening and conversation tests. In a listening test, a group of people listens to a recording of an unmodified conversation and compares it with a simulation of the same conversation modified by the algorithm. This method can also be used to compare different algorithms. Any impairments caused by an algorithm are clearly audible, which makes listening tests a controlled means of evaluating algorithms.

Listening tests are particularly helpful during development, but must be augmented by conversation tests, in order to accurately evaluate impairment factors. User perception of clipping in a listening test may be more irritating than it would be in a real conversation, since the side tone in a genuine conversation masks short speech gaps. Moreover, a listening test is inadequate for evaluating residual echo: listeners may have difficulty perceiving recorded echo as genuine echo, because they do not hear their own voices and the echo is not coordinated with their own speech.

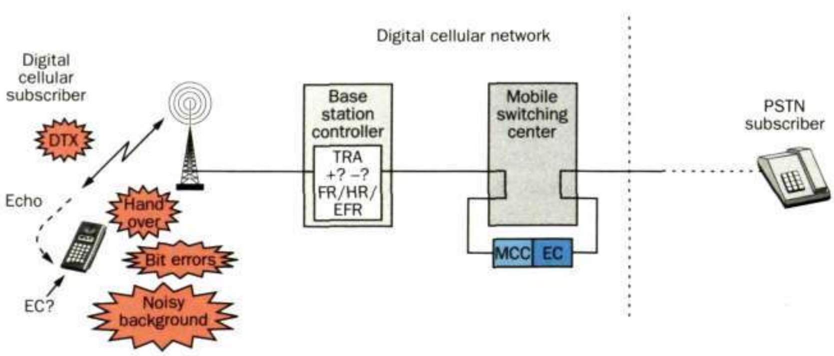
The evaluation of algorithm performance is indeed quite complex, and an algorithm for cancelling echo from the mobile handset end must satisfy many requirements. For example, the tests must evaluate the performance of an algorithm under several different situations (Figure 8):

- Different mobile handsets—different handsets produce different amounts of echo. Some mobile handsets have built-in echo cancellers, which also differ in performance. An algorithm for cancelling echo from the mobile handset end must never allow deterioration of speech quality, especially for a handset that does not produce echo.
- Different speech-coding algorithms—all speech-coding algorithms are not equally faithful in coding; superior speech-coding algorithms make the echo easier to hear. For example, the GSM enhanced-full-rate offers better speech coding than GSM full-rate. Consequently, the former should produce a clearer echo than the latter.
- Different background sound—because the handset is portable, it is used in environments with varying background sound, such as music or ambient noise in cars or buildings. When the algorithm activates attenuation, the comfort noise generator must be able to handle any of these situations.
- Different line-level settings—the level of echo depends on the line level. For example, if the line levels in the transcoder are changed, the level of the echo will also change; if the line level on the downlink is increased, the level of the echo will also increase.
- Different handset loudspeaker volumes—the level of the echo depends on the volume setting of the handset. If the loudspeaker volume is increased, the level of the echo increases.
- Various bit error rates—bit errors in the radio interface make echo non-linear, and the complete loss of frames causes time variations in the echo path.
- Handover—time variations are introduced at handover, owing to the loss of speech frames. The echo delay might also change when the handover is between base stations that use different hardware.
- Discontinuous transmission—when the DTX function is activated in the uplink, spurious echo bursts may be transmitted.

The MCC algorithm has been tested for each of the above situations. Mobile crosstalk control cancels echo, if present, without clipping or distorting speech from the mobile handset end, and without inserting faulty comfort noise when background sound is present. Hence, the MCC algorithm successfully enhances speech quality in digital cellular networks.

Figure 8
System characteristics that influence MCC performance. GSM shown as an example.

TRA	**Transcoder**
FR	**Full-rate**
HR	**Half-rate**
EFR	**Enhanced full-rate**
+?–?	**Line levels differ**



Echo cancellers with mobile crosstalk control

Ericsson's family of echo cancellers in pool (ECP)[2] products has proven effective in a variety of systems. Now, mobile crosstalk control is available in a digital signal processor (DSP) software module designed to be loaded on existing ECPs. Operators

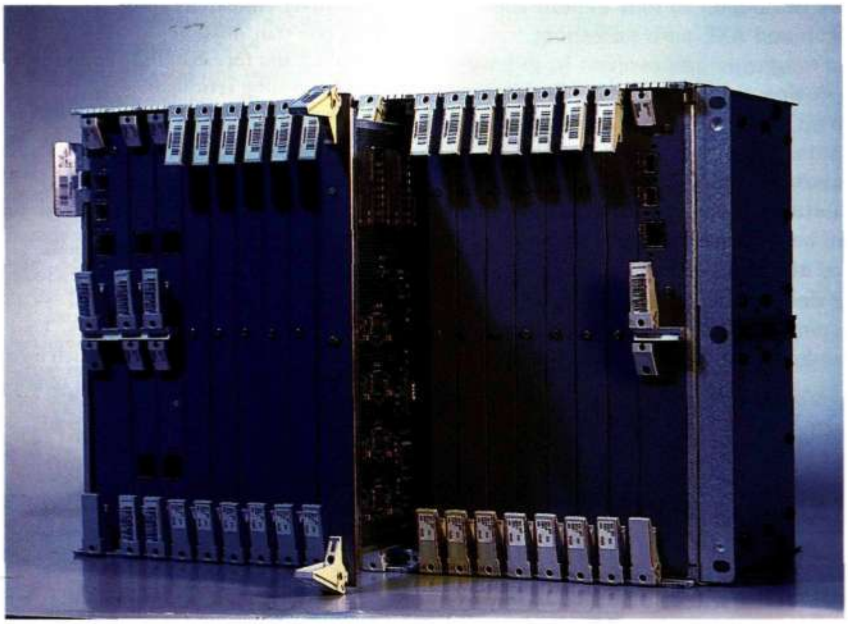


Figure 9
ECP 424: MCC implemented on the ECP 404 hardware platform. A single board contains 32 echo canceller devices featuring MCC.

already using echo cancellers in pool can add mobile crosstalk control without altering their current hardware platform or changing the number of echo cancellers in their exchange cabinets. Ericsson also offers new products that include mobile crosstalk control (Figure 9):

- ECP 323 (the ECP 303 with MCC);
- ECP 424 (the ECP 404 with MCC).

The MCC function in the ECP 323 and ECP 424 echo cancellers is always active in traffic cases that require a network echo canceller, thereby providing extra enhancement of speech quality. Also, because of the high degree of variation in the path delay of the echo from the mobile handset end, MCC adapts to any delay between 144 and 320 ms, with no parameter adjustment needed. Finally, the algorithm of the network echo canceller part is unchanged in the new units and thus continues to provide the same excellent speech quality as demonstrated in independent evaluations.

Conclusion

Fundamental differences in echo that originate in the PSTN local loop (network echo) and mobile digital handsets (acoustic crosstalk echo) require different echo cancellation methods. Ericsson's MCC software estimates the delay and level of echo in mobile handsets, takes into account the diverse situations that can cause the occurrence and variation of echo, and successfully attenuates it, yielding improved speech quality.

The MCC software augments existing software in Ericsson's ECP products without affecting the ECP's algorithms for cancelling network echo. The new ECP combinations thus provide enhanced speech quality in digital cellular networks.

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GDM-based generation of AXE core switching devices

Jan Hopfinger and Björn Sundelin

The generic device magazine concept was developed to fully exploit the advantages of the rationalized group switch and AXE core switching devices. The idea of a standard, equipped magazine has evolved to promote a flexible and economical use of generic device magazines. The advantage of GDMs is that they can be fully assembled and tested at the factory. They can then be mounted into standard cabinets which, when fully assembled, can also be tested in-house. As a consequence, entire nodes can be assembled and tested before delivery. To the customer, this means that the duration of on-site installation and testing can be reduced from several weeks to as little as one day, depending on the size of the node.
The authors describe AXE core switching devices, explaining the evolution of exchange terminal circuits and introducing the latest generation of Ericsson's tone-handling and signaling products, including the PDSPL-2H, RMS and no. 7 signaling terminals.

AXE core switching devices

AXE core switching products fall into two different categories: products that belong to the group switch subsystem (GSS) and AXE core switching devices, which consist of APT devices and extended switching subsystem (ESS) products. The new group switch was described in Ericsson Review no. 2, 1997. Our article covers a similar evolution of AXE core switching devices.

Two salient features of AXE core switching devices may be pointed out. They are connected to the group switch—via digital link interfaces—and they belong to the APT part of AXE. Today, core switching devices fall into four families:

- Extension terminal (ET) products for primary rate transmission.
- Signaling and tone-handling equipment.
- No. 7 signaling terminals.
- Announcement service terminals (AST).

To these, two new families will soon be added; namely,

- high-speed extension terminal products for 155 Mbit/s transmission;
- high-speed no. 7 signaling terminals.

Additional core switching devices include: pulse-code devices (PCD), digital PCD (PCD-D) and equipped magazines. Core switching devices do not comprise their own subsystem within AXE, but belong to several subsystems, such as

- the trunk and signaling subsystem (TSS);
- the subscriber switching subsystem (SSS);
- the operation and maintenance subsystem (OMS);
- the common channel signaling subsystem (CCS);
- the extended switching subsystem;
- the remote measurement subsystem (RMS).

System evolution

The current generation of AXE core switching devices has evolved around four key factors:

- digital link interface 3 (DL3);
- generic device magazines (GDM);
- smaller footprint;
- hardware platforms.

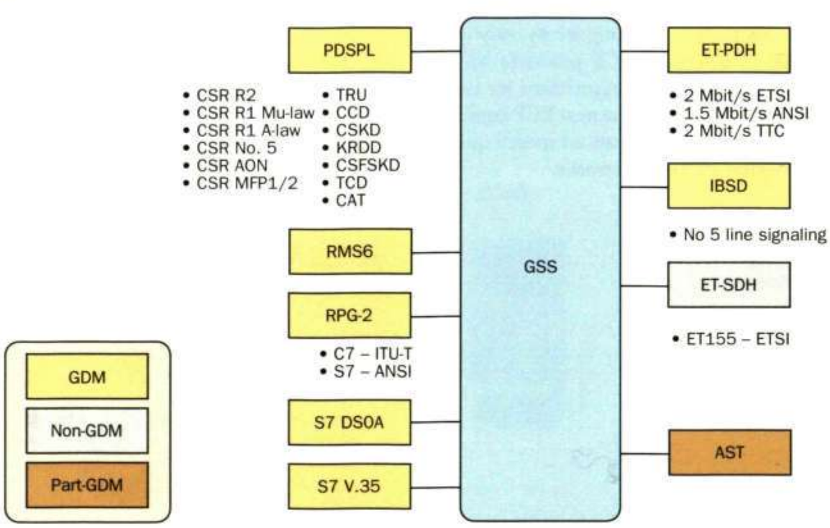
Together, these factors yield the largest evolution of core switching devices since AXE was created in the mid-1970s.

DL3 interface and the GDM concept

The introduction of the new internal DL3 interface constituted a big improvement in the AXE core switching part. The DL3 succeeds the digital link 2 (DL2). Serving as an internal interface that carries up to 512 speech channels, it carries sixteen times as much traffic per cable.

The DL3 interface greatly reduces internal cabling and also makes it possible to mix traffic from different devices in the interface to the time switch module (TSM) part in the switching subsystem. The generic device magazine concept was developed to fully exploit the advantages of the DL3. A generic device magazine is a magazine whose architecture permits different devices to be mixed/mounted very freely (Figure 2).

Figure 1
APT devices.



Generic device magazines

Each generic device magazine consists of

- one pair of regional processors (RP);
- one pair of digital link multiplexers;
- sixteen device slots;
- a card cage and the backplane.

The new regional processor (known as the RP4) manages all regional software functionality for extension module (EM) device boards within a given generic device magazine. According to the GDM concept, a pair of regional processors is co-located on the magazine with the devices it owns. This arrangement minimizes the amount of cabling needed between regional processors and devices.

The connection between each pair of RP4s and the device slots (EM bus) is fully implemented in the backplane. A maintenance bus between regional processors and devices has also been implemented in the backplane. The maintenance bus supports board identification and indication functions. Additional functions on the RP4 boards supervise power and power distribution.

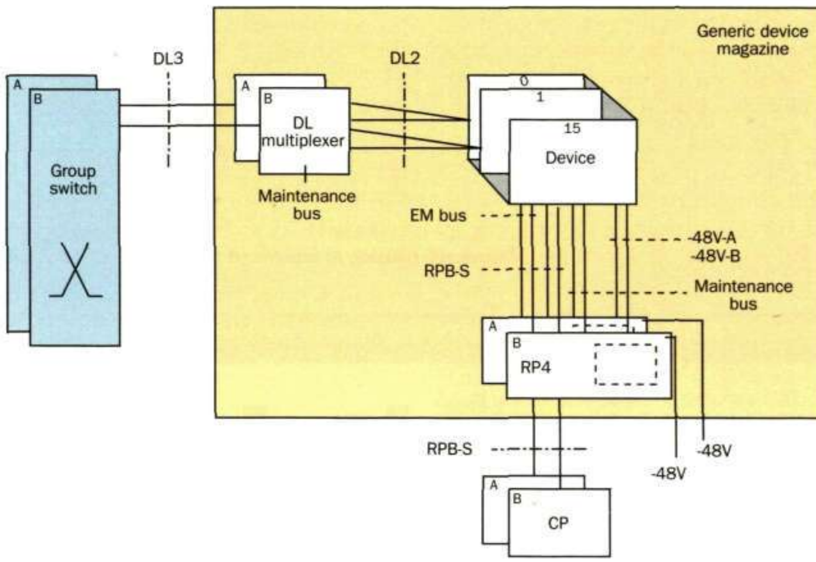


Figure 2
Hardware architecture of the generic device magazine (subrack).

Box A
Abbreviations

ANSI	American National Standards Institute
ASIC	Application-specific integrated circuit
AST	Announcement service terminal
BSC	Base station controller
CAS	Channel associated signaling
CAT	Code answer and tone sender
CCD	Conference call device
CCS	Common channel signaling (subsystem)
CMOS	Complementary metal-oxide semiconductor
CRC	Cyclic redundancy check
CSK	Code sender for DTMF tones
CSR	Code sender/receiver
CSR R2	Code sender/receiver, signaling system R2
DL2, DL3	Digital link 2, digital link 3
DP	Device processor
DRAM	Dynamic RAM
DSOA	Digital signal (level 0) A
DSP	Digital signal processor
DTMF	Dual-tone multifrequency
EM	Extension module
ESS	Extended switching subsystem
ET	Exchange terminal
ETC	Exchange terminal circuit
ETSI	European Telecommunications Standards Institute
GDM	Generic device magazine
GDM-F	Full-size GDM
GDM-H	Half-size GDM
GRETA	Group switch and exchange terminal adapter
GSS	Group switch subsystem
IOG	Input/output (I/O) group
IOG11	I/O system 11
IOG20	I/O system 20
ISDN	Integrated services digital network
ITU-T	International Telecommunication Union–Telecommunications Sector
KRD	Keyset receiver device
MSC	Mobile switching center
NTT	Nippon Telegraph and Telephone Corporation
OC-1	Optical carrier, 51 Mbit/s link
OC-3	Optical carrier, 155 Mbit/s link
OMS	Operation and maintenance subsystem
PCD	Pulse-code device
PCD-D	Digital PCD
PCI	Peripheral component interconnect
PCM	Pulse code modulation
PDSPL	Pooled digital signaling platform
PDSPL-2H	Second generation, half-size PDSPL
PLL	Phase-locked loop
PRA	Primary rate access
PROM	Programmable read-only memory
RAM	Random access memory
RMS	Remote measurement subsystem
RMS6	Sixth-generation remote measurement subsystem
RMS6-F	Sixth-generation full-size RMS
RP	Regional processor
RP4	Fourth-generation RP
RPB-S	Serial RP bus
RPG	RP with group switch interface
RPG-2	Second-generation RPG
RPP	RP with PCI interface (RPP is used for datacom applications)
SCSI	Small computer system interface
SDH	Synchronous digital hierarchy
SDI	Serial device interface
SP	Support processor
SSM	Synchronization status message
SSS	Subscriber switch subsystem
STP	Signal transfer point
T1	Physical 1.5 Mbit/s link, ANSI standard
TCD	Trunk continuity check device
TSM	Time switch module
TRH	Transceiver handler
TT	Test telephone
TTC	Telecommunications Technology Council (Japan)
TSS	Trunk and signaling subsystem
VLSI	Very large-scale integrated circuit

Box B
Equipped GDM magazines

Several versions of equipped GDMs exist. Three of them are presented below.

Example 1
GDM-H equipped as follows:

Item	*Quantity*
ETC 5	14
RPG-2	1

This is a high-volume configuration used in MSCs for GSM and other configurations as well as transit and local exchanges. To avoid blocking in the GSS, one RPG-2 (used as an STC or no. 7 or TRH signaling terminal) is placed in the GDM-H together with fourteen ETC 5s.

Example 2
GDM-H equipped as follows:

Item	*Quantity*
CAT	1
KRD	1
CCD	1
TCD	1
CSR R2	2
RPG-2	2

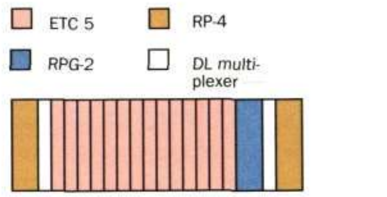
This configuration is mainly used in MSCs for the AMPS/D-AMPS market. Tone and signaling equipment are shown mixed with RPG-2 in the same magazine.

Example 3
GDM-H equipped as follows:

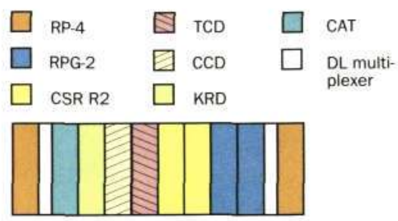
Item	*Quantity*
CCD	1
CSK	1
CSR R2	1
ETC 5	1
RPG-2	3

Three spare slots may be used for extra equipment; for example, for a test phone. This configuration is mainly used in the mini-MSC for GSM. ETCs, tone and signaling equipment and RPG-2 can be mixed.

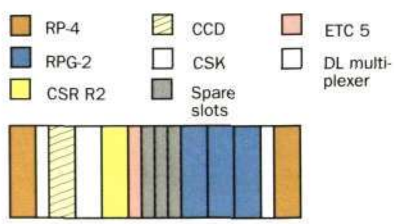
Note: Actual board placement may vary.



Example 1



Example 2



Example 3

As with the regional processors, the digital link multiplexers have been duplicated. They multiplex sixteen DL2 interfaces into a single DL3 interface for each switch plane. This reduces the amount of cabling, since all DL2 interfaces in the generic device magazine are implemented in the backplane. In short, sixteen DL2 cables have been reduced to just two DL3 cables per generic device magazine.

In the backplane, a serial RP bus (RPB-S) is connected to each device slot. The connection enables extension modules and regional processors—such as the BYB 501-version regional processor with group switch interface (RPG-2)—to be mounted in the same magazine. Moreover, it ensures that the generic device magazine can accommodate future device boards with on-board regional processors.

Each generic device magazine contains sixteen device slots, the use of which permits device boards to be mixed very freely. Some dimensioning restrictions apply, however. For example, physical space is limited and the capacity of the regional processors must match the needs of the boards in the generic device magazine. Notwithstanding, the GDM concept offers new, previously unattainable possibilities. In terms of regional processor capacity, for instance, the unused slots of a GDM equipped with devices that consume a great deal of capacity can be equipped with devices that consume little or no RP4 capacity (such as no. 7 signaling terminals on the RPG-2 platform). The code sender/receiver R2 (CSR R2) is one example of capacity-hungry devices, where each board requires one-fourth of the paired regional processor capacity.

Equipped GDMs

The concept of a standard, equipped GDM has been developed to promote a flexible and economical use of generic device magazines. Initially, standard cabinets were created on the basis of requirements for standard configurations of complete nodes; for example, transit nodes, local nodes, signal transfer points (STP), mobile switching centers (MSC) and base station controllers (BSC). Proceeding from the standard cabinets, designers next created several standard, equipped generic device magazines (Box B).

The advantage of having standard GDMs is that they can be fully assembled and tested at the factory. These, in turn, can be mounted into standard cabinets which, when fully assembled, can also be tested in-house. Finally, the entire node can be assembled and tested before delivery. Thus, standard generic device magazines make up the cornerstone of node-manufacturing centers. To the customer, this means that the duration of on-site installation and testing can be reduced from several weeks to as little as one day, depending on the size of the node.

Another advantage of standard generic device magazines is that engineers can use them to optimize the design of cabinets and nodes. Because components can be mixed very freely in generic device magazines, engineers can design cabinets and nodes around the magazines, making use of empty space and eliminating unnecessary hardware.

Two different versions of the generic device magazine currently exist:

- The GDM-H for half-size boards; that is, for 115 x 175 mm BYB 501 boards;
- The GDM-F for full-size boards; that is, for 265 x 175 mm BYB 501 boards.

Any of the following devices can be mounted in the GDM-H:

- The latest ETC 32, known as the ETC 5.
- The latest ETC 24, known as the ETC-T1H.
- Any pooled digital signaling processor platform (PDSPL) application on the PDSPL-2H platform.
- New signaling terminals on the RPG-2 platform.
- V.35 ANSI signaling terminal.
- DS0A ANSI signaling terminal.

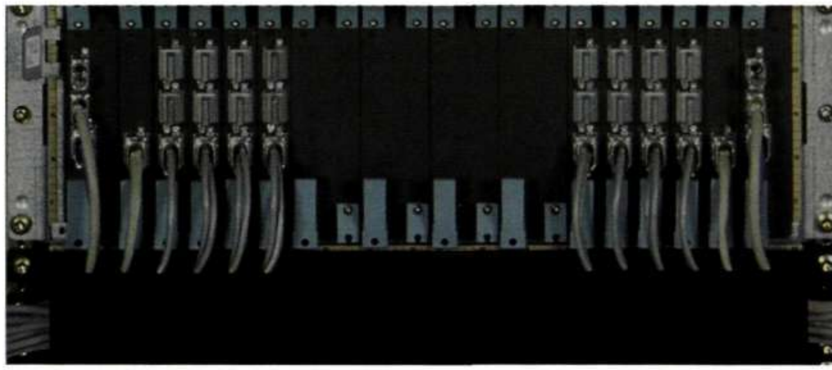


Figure 3
Equipped GDM-H magazine. The leftmost and rightmost boards are RP4s. Beside them are digital link multiplexers. The magazine contains eight ETC5s and four RPG-2s. This configuration is used mostly in base station controllers for GSM.

Box C
Transmission hierarchies

ETSI standard

64 kbit/s
One speech channel. Resulting 8-bit sampling of analog speech signal with sampling frequency 8 kHz. A-law coded. ETSI denominations are P0 = logical 64 kbit/s; E0 = electrical P0.

2,048 kbit/s
Primary rate in ETSI standard, 32 channels. Contains 30 speech channels, one administrative time slot and one signaling (CAS) time slot. In some applications (CCS), the time slot can also be used as a speech channel. P12s = logical 31 x 64 kbit/s plus TS0; E12 = electrical, HDB3 coded P12s.

155 Mbit/s *STM-1*
Lowest level in the SDH hierarchy. Contains 63 P12s and other information, mostly for monitoring transmission.

ANSI standard

64 kbit/s *DS0*
One speech channel. Resulting 8-bit sampling of analog speech signal with sampling frequency 8 kHz. Mu-law coded.

1,544 kbit/s *T1/DS1*
Primary rate in ANSI standard, 24 channels. Contains 24 speech channels, plus an 8 kbit/s channel for administering and monitoring transmission. Robbed bit signaling is used (CAS); otherwise, one of the speech channels may be used for signaling purposes (CCS).

51 Mbit/s *OC-1*
Lowest level in the SONET hierarchy. Contains 28 DS1s and other information, mostly for monitoring transmission.

155 Mbit/s *OC-3*
Next level in the SONET hierarchy. Contains 84 T1s and other information, mostly for monitoring transmission.

Japanese standard

64 kbit/s
One speech channel. Resulting 8-bit sampling of analog speech signal with sampling frequency 8 kHz. Mu-law coded.

2,048 kbit/s
Primary rate in Japanese 32-channel system. This is an NTT interface.

1,544 kbit/s
T1 (almost the same as in ANSI standard) is also used in Japan.

51 Mbit/s *STM-0*
Lowest level in the TTC SDH hierarchy.

155 Mbit/s *STM-1*
Next level in the TTC SDH hierarchy.

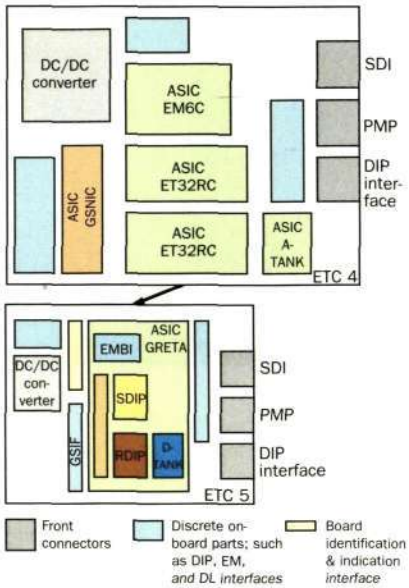


Figure 4
Evolution of the ETC 4 into the ETC 5.

A-TANK	**Analog tank**
DC/DC	**Direct current/direct current**
DIP	**Digital path**
DL	**Digital link**
D-TANK	**Digital tank**
EM6C	**Extension module circuit**
EMBI	**Extension module block interface**
ET32RC	**ET 32 receiver circuit**
ET32TC	**ET 32 transmit circuit**
GRETA	**Group switch and exchange terminal adapter**
GSIF	**Group switch interface**
GSNIC	**Group switching network interface circuit**
PMP	**Physical monitor points**
RDIP	**Receive DIP**
SDI	**Signaling device interface**
SDIP	**Send DIP**

- In-band signaling device.
- Test telephones (TT).
- Regional processors with PCI interface (RPP, used in applications that do not require Ethernet).

The objective is to design all core switching devices as half-size boards in the GDM-H (apart from low-volume products, such as the RMS6 and ETC J32).

The following devices can be mounted in the GDM-F:

- First-generation ETC 24, ported into BYB 501, ETC-T1F.
- ETC 32 (for Japan), ported into BYB 501, ETC J32.
- RMS in BYB 501, RMS6-F.

Exchange terminal circuits

Exchange terminal circuits (ETC) are interfaces between AXE and the external transport network. They can be found in almost all AXE nodes, including international, transit and local exchanges, MSCs and BSCs. In the world market, three major transmission standards exist for primary-rate pulse-code modulation (PCM) links:

- 2,048 kbit/s ETSI standard.
- 1,544 kbit/s ANSI standard.
- 2,048 kbit/s Japan-specific standard.

Exchange terminal circuits have been developed for each of the standards and introduced in the BYB 501.

ETC 32

The ETC 32—the exchange terminal circuit that complies with the ETSI standard—is a high-volume product of which several hundred thousand examples are manufactured each year. The huge market volumes have made it necessary to improve and rationalize the product several times. The latest version, which has been introduced in the BYB 501, is called ETC 5.

The ETC 5 was designed to be a single, half-size board for the GDM-H. It is only half as large as its predecessor, the ETC 4. Its diminutive size was achieved mainly by reducing the number of application-specific integrated circuits (ASIC) from five, in the ETC 4, to one in the ETC 5 (Figure 4). Combined with advanced board design

Box D, New functions in the ETC 5

The ETC 5 can be set in two different modes:
- Fully software-compatible with the previous ETC 32; that is, the ETC 4 (called ETC 4-mode).
- With new functions (called ETC 5-mode).

In addition to functions deployed in earlier generations of the ETC 32, several new functions exist in ETC 5 mode:
- New slip supervision, including settable hysteresis. Slip hysteresis has been implemented on the basis of new synchronization requirements.
- Automatic CRC-4 mode. The ETC 5 can automatically select CRC-4 or non-CRC-4 mode, depending on whether or not the equipment in the other end supports CRC-4.
- A-law/Mu-law conversion, ability to send more tones, attenuation and Mu-law-coded tones. Five different tones can be sent simultaneously. Note: the "tone bank" contains more than five tones.
- Selectable idle-code patterns. It is possible to choose between idle code and quiet code.
- ISDN primary rate access (30B + D), V3 layer 1 interface. Sectioned maintenance is supported.
- ETSI V5.2 layer 1 capability. V 5.2 is a standard interface between local exchange and access node.
- SSM. Synchronization status message is supported by the ETC 5 hardware.
- Board identification and indication functionality. Board identification and indication is a standard feature of all GDM-based devices.

using 3.3V components, the new ASIC (called GRETA) lowers power dissipation to 0.8W per board.

GRETA, which stands for *group switch and exchange terminal adapter*, accounts for most of the recent hardware evolution, and is the realization of a 0.5 μm complementary metal-oxide semiconductor (CMOS) process from Texas Instruments. It contains integrated on-chip random access memory (RAM) and 40,000 used-gate equivalents. The integration yields great savings in terms of board space, power dissipation and increases reliability.

In addition to its smaller size, the ETC 5 also features several new functions, the most important being

- automatic cyclic redundancy check (CRC-4) mode;
- integrated services digital network primary rate access (ISDN PRA) V3 layer 1 interface;
- ETSI V5.2 layer 1 capability (Box D).

The ETC 5 is also backward-compatible with the ETC 4.

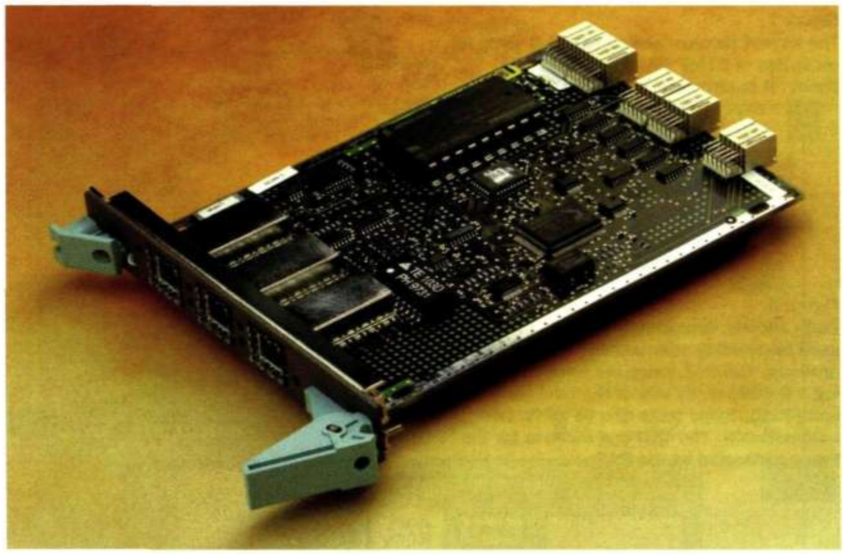


Figure 5
Latest version of the 32-channel ETC, ETC5.

ETC 24

The ETC 24 complies with the ANSI standard for exchange terminal circuits. It is used in North America, Egypt, and certain countries in Southeast Asia (Hong Kong, South Korea and Japan).

ETC-T1F

The ETC 24 has a long history of evolution. In 1986, it was implemented in a single-board solution.[1] Later, in 1996, a new ETC 24—called the ETC-T1—was introduced with support for monitoring performance and functionality for the Japanese CRC. As introduced in the BYB 501, the ETC 24 is very similar, in terms of functionality, to the ETC-T1. Designers have added board identification and indication functionality (a standard feature in the generic device magazine) and excluded the ETC 24/96, thereby creating a pure ETC 24 structure.

ETC-T1H

Merely porting such an important product as the ETC 24 to the BYB 501 was not enough, however. Instead, engineers have continued to refine its design. The latest version, named ETC-T1H, is contained on a single, half-size board in the GDM-H. The ETC-T1H integrates into one ASIC the functionality that had previously been implemented in nine separate circuits on the ETC-T1F. Furthermore, power dissipation

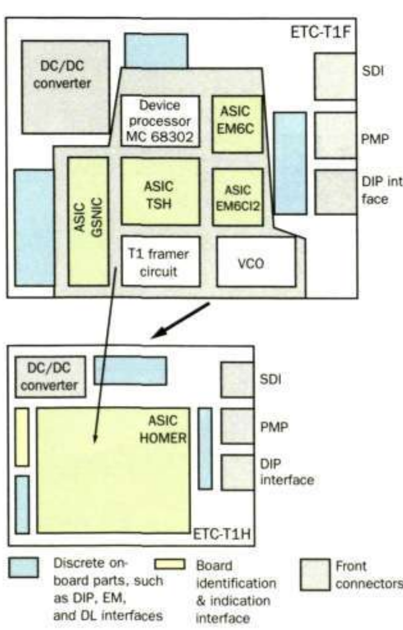


Figure 6
Evolution of the ETC T1F into the ETC T1H.

DC/DC	**Direct current/direct current**
DIP	**Digital path**
DL	**Digital link**
EM6C	**Extension module circuit**
GSNIC	**Group switching network interface circuit**
PMP	**Protected monitor points**
SDI	**Signaling device interface**
VCO	**Voltage controlled oscillator**

Figure 7
The keyset receiver device (KRD) is used for reception of DTMF digits from a keyset phone. It is also used for sending dial tone.

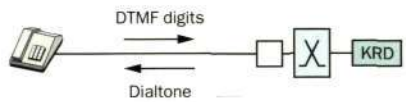


Figure 8
Code answer, tone sender device (CAT) is used for sending maintenance tones or code answers. The CAT tones are invoked by calling a B-number. By use of the tone-receiving unit (TRU), these tests can be performed automatically. The TRU is a receiver for the tones generated by the CAT.

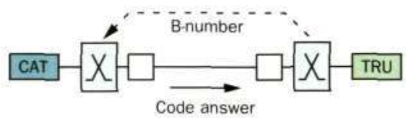


Figure 9
Several tone-based systems for interexchange signaling are implemented on the PDSPL platform; for example, R1, R2 and no. 5 (ISBD is a special board used for no. 5 line signaling).

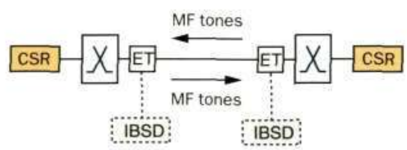


Figure 10
The tranceiver continuity check device (TCD) is used for verifying the speech path before a call is set up. It is used in some cases of common channel signaling.

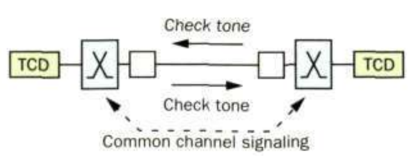
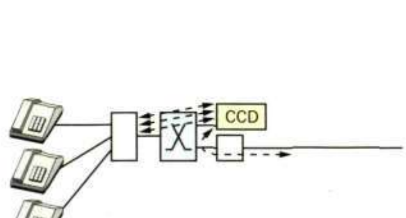


Figure 11
The conference call device (CCD) is used for conference calls and other functions where several speech paths are interconnected.



has been reduced by 50%. The hardware of the ETC-T1H was designed for two new functions; namely,

- the primary rate access interface, at reference point T;
- the handling of synchronization status messages (SSM).

Nearly all functionality in the ETC-T1H has been integrated into a single ASIC. HOMER, as it is called, is a 0.35 μm CMOS gate array developed using very large-scale integrated circuit (VLSI) technology. The T1 framer is handled by a frame block, which is a sourced macro at the Verilog net level. The device processor—a Z80 processor core—was also sourced at the Verilog net level and integrated into HOMER. All told, HOMER contains 75,000 gates and 34 kilobytes of RAM. A PLL (phase-locked loop) circuit has also been integrated into the ASIC.

The large-scale integration made it possible to produce the ETC-T1H as a half-size board in the GDM-H.

Japanese ETC 32

The ETC J32, which was developed for the Japanese standard, is solely used in mobile applications in Japan. In the BYB 501, it has the same functionality as its predecessor with the addition of board identification and indication functionality.

New tone-handling and signaling products

PDSPL-2H

In AXE, several functions exist for sending and receiving tones. They send and receive DTMF tones, interexchange signaling tones, and maintenance-related tones.

These functions are currently implemented in AXE on a variety of hardware platforms that range from analog equipment connected via pulse-code devices to modern digital signal processor-based (DSP) equipment. In some cases, one PDSPL-2 board can replace up to two cabinets of existing equipment.

A new generic hardware platform has been developed that can be used for each of these functions—the PDSPL-2H, which stands for *second generation, half-height, pooled digital signaling processor platform*.

The PDSPL board, which is half the height and double the width of a regular board, contains

- a device processor, which handles com-

munication to the RP4 processors in the GDM-H magazine;
- three digital signal processors, which execute algorithms for sending and receiving tones.

The three digital signal processors are connected to a 2 Mbit/s time-slot bus that enables them to be freely allocated for sending data to, or receiving data from, any time slot.

Firmware is stored in flash programmable read-only memory (PROM). When reset, the device processor part is copied into dynamic RAM (DRAM) and the DSP part is downloaded into internal memory. Two ASICs are used for interfacing the microprocessor with the EM bus. A third ASIC is used for interfacing the time slot bus with the group-switch DL2 interface in the GDM-H backplane.

The applications that run on the PDSPL were designed on top of a firmware platform that contains functions for inter-processor

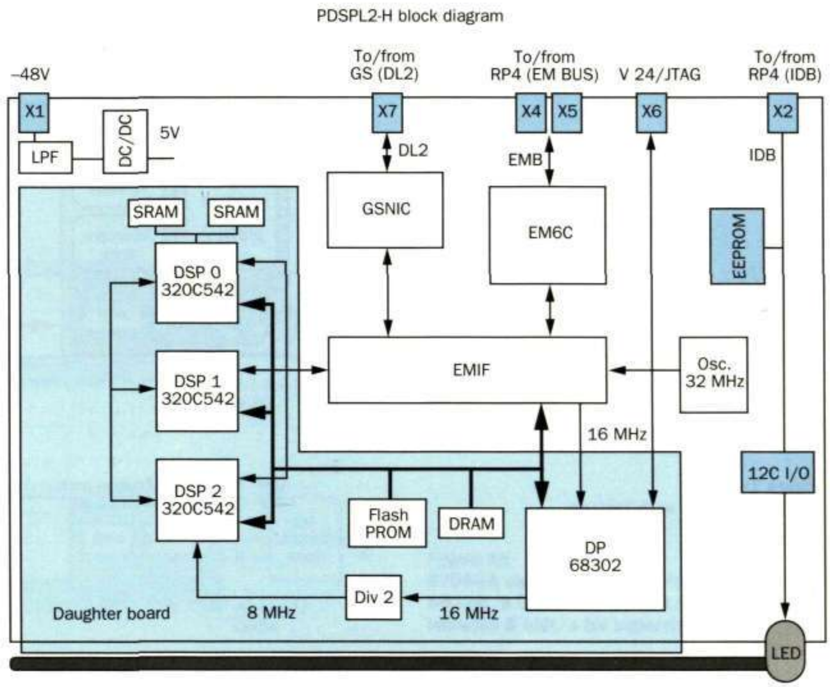
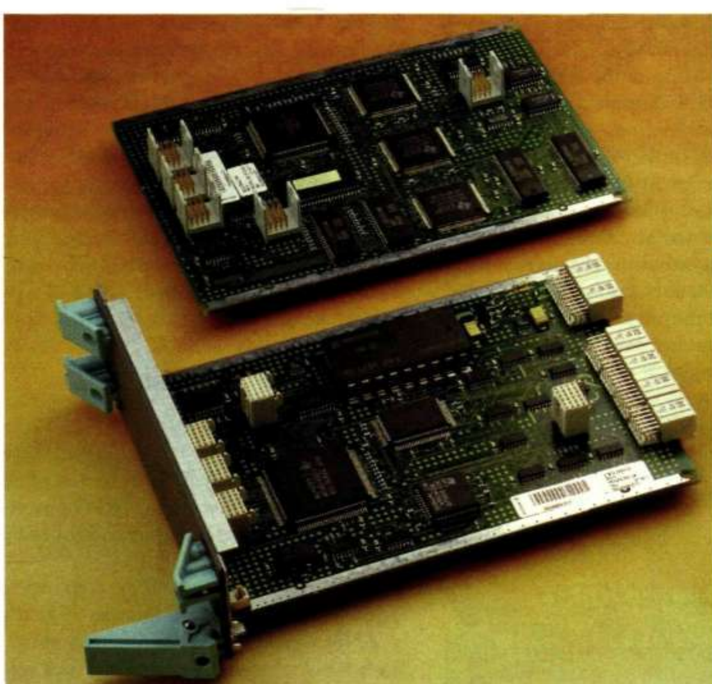


Figure 12
PDSPL hardware block diagram.



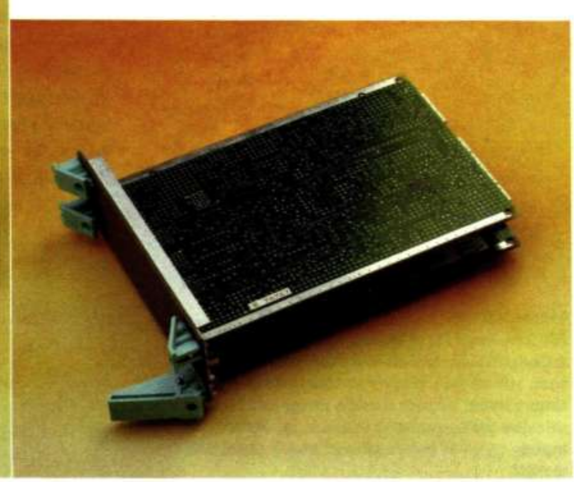


Figure 13
Photograph of the PDSPL-2H, a double-width, half-size board to be placed in the GDM-H.

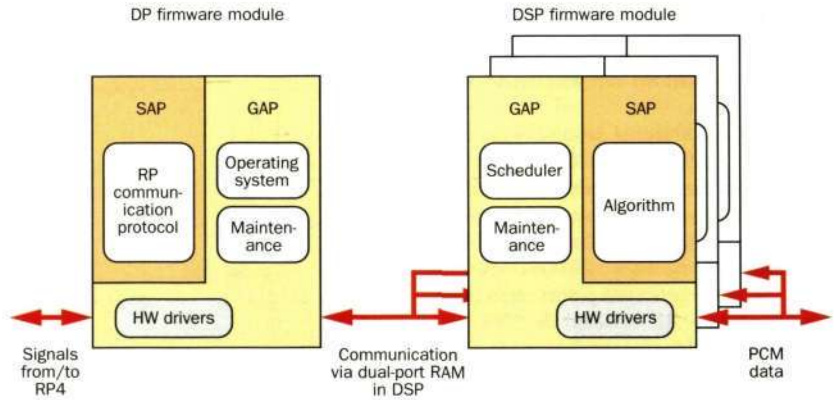


Figure 14
PDSPL firmware architecture.

communication and maintenance. Apart from time-critical DSP parts, which have been coded in assembler language, the firmware was written in C.

RMS

The remote measurement subsystem (RMS) makes end-to-end measurements for routine and diagnostic tests of speech path transmission quality.

The functionality of the RMS6 is similar to that of PDSPL-2 applications, with two main differences:

- The RMS6 contains a new hardware board, since it requires more capacity than the PDSPL-2 platform provides. The board contains eight DSPs.
- Remote measurement subsystem software is situated in the support processor (SP) in the input/output group (IOG)—unlike software for other devices, which is located in the regional or central processors. The RMS6 works with the I/O system 11 (IOG11) as well as with the new I/O system 20 (IOG20). The physical interface has also been improved, using X.25 instead of SCSI. X.25 facilitates a more flexible physical placement of RMS hardware within the node.

The RMS6 is a full-size board in the GDM-F (RMS6-F).

In-band signaling device

An in-band signaling device that can perform CAS no. 5 line signaling has been introduced in the BYB 501. It is a half-size board situated in the GDM-H. Each in-band signaling device is connected to an ETC 32 via a signaling device interface.

The main parts of the no. 5 signaling device are:

- a digital signal processor;
- a device processor (DP);
- interface circuitry.

New no. 7 signaling terminals

Signaling terminals for no. 7 signaling have been implemented in AXE in two ways: as regional software that runs on RPG processors and as dedicated hardware controlled by RP4 processors.

The RPG-2 based signaling terminal is used for ITU-T signaling links at 64 kbit/s or ANSI signaling links at 56 or 64 kbit/s. Each RPG is capable of handling four signaling links. Since the RPG is connected to

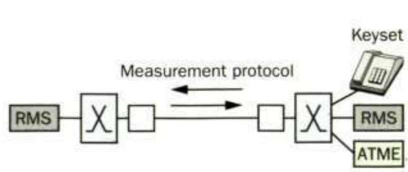


Figure 15
RMS is used for transmission quality measurements on end-to-end connections. The tool, which can be driven manually and by time tables, covers a wide range of ITU transmission measurement standards. RMS is mainly used in international and mobile networks.

the group switch, the signaling links are connected, semi-permanently, to the ETCs and are therefore accessible as time slots in either a 1.5 Mbit/s or 2 Mbit/s pulse code modulation system.

The signaling terminal for an ANSI 56 kbit/s link with a V.35 interface has been implemented as a dedicated half-height, double-width board. The V.35 signaling terminal fits in the GDM-H magazine. The V.35 link is accessible via a connector on the board front.

The signaling terminal for an ANSI 56 kbit/s link with DS0A has been implemented in a similar fashion; that is, as a dedicated board in the GDM-H. The signaling terminal has no group switch connection. The signaling link is accessible from the board front.

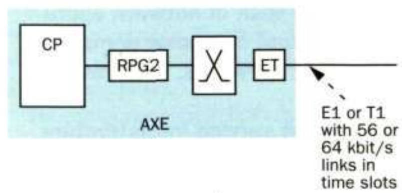


Figure 16
RPG2-based ITU-T or ANSI signaling terminal.

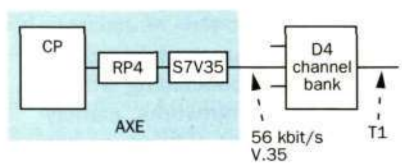


Figure 17
S7V35 signaling terminal for ANSI 56 kbit/s links.

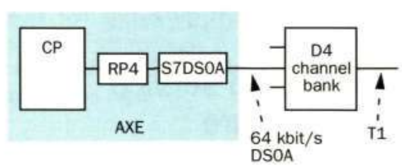


Figure 18
S7DS0A signaling terminal for ANSI 56 kbit/s links. The 64 kbit/s DS0A includes 8 kbit/s for supervision.

Future evolution

The evolution described in this article solely outlines the beginning of development and refinements in AXE core switching devices.

Further evolution within the family of extension terminals will introduce high-speed synchronous ETCs, starting with the ET 155, which will be ETSI-compliant.

Conclusion

The introduction of DL3 interfaces between the group switch subsystem and device magazines, along with the introduction of the GDM concept, has greatly improved flexibility in magazine, cabinet and node design, resulting in better usage of the group switch and a reduction of the physical hardware footprint.

Likewise, the introduction of flexible hardware platforms, such as the RPG-2 and the PDSPL-2, has reduced the number and physical size of hardware. Another benefit of this evolution is significantly lower power dissipation.

Finally, the concept of a standard, equipped magazine makes it possible to assemble and test complete GDMs in the factory. These, in turn, can then be mounted into standard cabinets, which can also be tested in-house. Indeed—thanks to the GDM concept—entire nodes can be assembled and tested before delivery. Thus, depending on the size of the node, this simple concept can save customers considerable time and money by reducing the time for on-site installation and testing.

Reference

1 Mårtensson, L., Nyström, B. and Samuelsson, A.: New exchange terminal circuit and MF signalling equipment in AXE 10. Ericsson Review 64 (1987):2, pp. 74-84.

Telecom management as a competitive tool

Kjell Andersson, Ingemar Häggström, Jan Insulander and Tomas Rahkonen

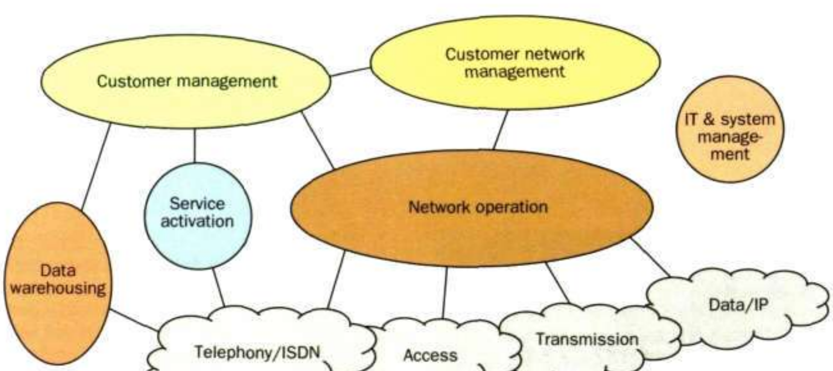
In the business-oriented telecom environment, the task of network operators and service providers is to satisfy subscriber and customer demands. Therefore, the systems and methods for managing networks, services and customers play a crucial role.
Efficient implementation of a telecom management service infrastructure helps reduce costs, improve quality and speed up the delivery of service. For new operators, this means short time to business and rapid payback on investments. For established operators, it means optimizing network operation and streamlining the organization, thereby strengthening overall competitiveness.
The authors describe the six management-application areas of Ericsson's telecom management products and services portfolio, concluding with a close-up look at three integrated solutions for telecom networks, namely network traffic management, customer management and network operation and maintenance.

Customized service infrastructure

A customized service infrastructure—which covers complete end-to-end operator processes—is being built using Ericsson's telecom-management solutions. Efficient solutions connect the physical network infrastructure to customer care and billing. Consequently, the new solutions bridge the gap between business support and operations support and remove obstacles that prevent successful business development.

Ericsson offers enabling technologies and methods for both new and incumbent operators. The solutions comprise a new generation of telecom-management solutions, customized software components and system-integration services.

Figure 1
The telecom management application framework comprises six critical telecom management areas: network operation, customer management, service provisioning, data warehousing, IT and system management and customer network management. One or more best-of-breed components has been selected for each application within these areas, making it possible to support any scale operation. The products have been integrated to provide a complete working solution that covers the entire telecom management process.



Interaction optimizes in-service performance

The prerequisite of efficient telecom management is automated process flow-through. Ericsson's telecom-management solutions include services for engineering operator processes. These services—which use computers, not operator staff, as their primary building blocks—simplify processes, giving them fewer steps. Greater automation gives operator personnel and customers faster and easier access to relevant information. It also shortens response times, ensures consistent quality and reduces transaction costs.

Comprehensive application framework

Ericsson's commitment to telecom management comprises a comprehensive portfolio of products and services within six management-application areas. End-to-end process flow-through, which is ensured by means of cooperation and coordination between these areas, also provides support for service-assurance and service-provisioning processes. Ericsson's application framework comprises

- network operation—for the implementation of cost-effective operation and maintenance (O&M) throughout the telecom network;
- customer management—for recruiting and retaining profitable customers;
- service activation—for speeding up service-provisioning lead times from days to minutes through automatic service activation;
- customer network management—for controlling network resources and services that customers can exercise themselves;
- data warehousing—for decision-making support, based on input (such as call data, statistics and real-time information) from the network;
- information technology (IT) and system management—for administering management systems.

It should be mentioned that the application framework does not explicitly address planning tools, although these represent a strategically important area for Ericsson. Finally, the management-application areas provide for smooth mapping to the operator business-process model defined by the Network Management Forum (NMF).

From components to customized solutions

Ericsson's common application framework covers all telecom management domains and ensures interoperability within and between different areas of telecom management. The framework is divided into application areas. These in turn include a broad range of software components. Combining or integrating the different components creates customized solutions.

Pre-integrated solutions ensure rapid deployment and interoperability

Ericsson builds its integrated solutions—which emphasize application interoperability and scalability—out of *best-of-breed* platform-independent components from independent software vendors (ISV). To obtain the best possible solution, Ericsson cooperates with independent software vendors who provide the very best components available on the market. In accordance with this partnering strategy, Ericsson alleviates operators of the burden of managing multiple software suppliers.

The components from ISVs can be likened to toolboxes, each of which offers different basic applications. By means of toolkits, rapid-application design environments and telecom know-how, Ericsson adds value to software from independent software vendors and integrates the components to fit operator processes.

The basic system is integrated before any specific customer contracts are signed, and each solution is tested before it is incorporated into a customer solution. For operators, this minimizes the risk of malfunctioning systems and non-compatible interfaces and speeds up the implementation process. Furthermore, it permits operators to have advance knowledge of costs, performance and capacity.

Efficient systems integration

Based on operators' specific business needs, each solution is adapted to the unique environment of new or established telecom com-

Box A
Abbreviations

CDR	Call detail record
CMIP	Common management interface protocol
CORBA	Common object request broker architecture
EDI	Electronic data interchange
EHPT	Ericsson Hewlett-Packard Telecommunications
GUI	Graphical user interface
HTML	Hypertext markup language
IOA	Inter-operator accounting
IRP	Integration reference point
ISV	Independent software vendor
IT	Information technology
ITU-T	International Telecommunication Union – Telecommunications Standardization Sector
NMF	Network Management Forum
NTM	Network traffic management
O&M	Operation and maintenance
SNMP	Simple network management protocol
WWW	World Wide Web

Figure 2
Customer management is one of the six areas within Ericsson's telecom management application framework. Customer-management solutions help operators to deal with long-term interactions between customer relations, accounting, and service provisioning. Solutions are available for every variety and size of network, making it possible to target new as well as established operators.

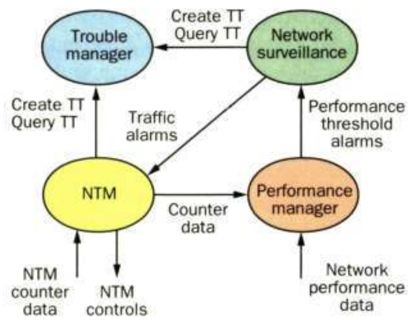


Figure 3
Interaction between components in a network traffic management solution.

panies. Each component, including Ericsson's added value, is integrated into the operator network and business environment, and new interfaces can be implemented to establish links between application areas. The implementation is facilitated using a broad range of new technologies and open interfaces.

- Established interfaces, such as the common management information protocol (CMIP) and simple network management protocol (SNMP), are used for communicating with network elements.
- Open standards, such as the common object request broker architecture (CORBA), transaction processing, and electronic data interchange (EDI) are used in the pre-integration process.
- Rapid-application development tools are used for optimizing software components.
- Java and the hypertext markup language (HTML) offer high productivity environments for designing and implementing graphical user interfaces (GUI).
- World Wide Web (WWW) browser-based applications and plug-ins, in combination with workflow engines, create excellent integration environments.
- Three-tier architectures simplify integration, especially in the data layer.

Pre-integration in practice

Network traffic management

New networking technologies, including broad bandwidth transmission, larger switches and more efficient signaling systems, make the telephone network increasingly vulnerable to overload and congestion. Unpredicted uptake and the use of innovative services contribute to the challenge of maximizing revenue while minimizing the impact that services have on network conditions. Examples of problems are mass calling and transmission failure. For instance, mass calling sometimes occurs in response to radio and TV programs, resulting in insufficient network capacity and widespread congestion.

Network traffic management is about maximizing call completion. This implies the need for up-to-date information on traffic situations and a means of controlling traffic when necessary—where controls mainly apply to restricting and redirecting traffic.

Traditionally, corrective network-traffic-management actions have been based on traffic measurements delivered by switching systems—typically in five-minute intervals. Due to the increased dynamics of modern networks, this data needs to be supplemented with information from other sources; for example, alarms and performance indicators from the signaling network and alarms from important transmission systems. There is virtually no limit to the number of information sources that are relevant to network traffic management.

The challenge is to devise a solution that is both flexible enough to meet future requirements for information management and mature enough for more or less off-the-shelf and short-notice delivery. In keeping with Ericsson's general strategy for providing solutions based on the integration of *best-of-breed* components, the network-traffic-management (NTM) solution has been devised to meet the requirements of a dynamic network environment.

The core of the solution is the NTM component, proven through several years of operation by Swedish Telia, collects five-minute intervals of data and provides the main display for the network traffic manager. It also provides the means of activating network traffic controls in the network.

Ericsson's network surveillance system complements the NTM solution with information on faults. Examples of information include alarms relating to equipment failure and conditions of excessive traffic load. Sources of alarms are switching systems, transmission systems, and signal-monitoring systems. If relevant to the NTM process, the alarms may be forwarded to the NTM component, indicating status via the NTM display.

Figure 4
Network operation is one of the six areas within Ericsson's telecom management application framework. To meet new demands, such as fixed-mobile convergence, multimedia, and data communication, modern networks require information from many different sources.

Also of relevance to the NTM process is information on past performance, historical trends, and predicted future performance. A performance-management system handles this task, using accumulated data from the NTM system as well as data reported directly from the switching systems.

The performance manager includes an alarm server that can send alarms to network surveillance functions when threshold criteria are exceeded. The performance manager thus complements network traffic management.

Another complement to network traffic management is the trouble manager, which is a general tool for reporting, tracking and following up network faults. In the context of network traffic management, the trouble manager enables the network manager to analyze a problematic traffic situation in terms of outstanding network problems, ongoing repairs, and so on. It also enables the network manager to create a trouble ticket for network conditions when manual repair or extensive reconfiguration is required.

Customer management

The deregulation of telecommunications represents a challenge to existing operators. New players are competing with established operators for market share. Areas of paramount importance are the rapid implementation of new services and adaptation to customer needs, which put new demands on customer-management systems, making them the tool for creating competitive edge.

Ericsson's customer-management solution, which is built around *best-of-breed* components of different functional areas, is integrated into a single virtual-system solution. The solution—which is comprehensive, both in depth and breadth of function—offers a common user interface for all its components.

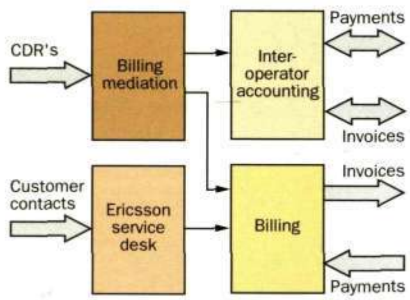


Figure 5
A basic customer-management system configuration.

Mediation device
Ericsson's mediation device collects and processes call detail records (CDR) from switches before it distributes the records to downstream systems. The flexible processing capabilities offered by the mediation device enable operators to charge for new services without introducing numerous changes in the billing system. The mediation device not only simplifies the management of growth and change in the network and services, but also

- supports the conversion of data between different formats;
- processes billing data;
- improves fraud-detection capabilities;
- improves the distribution of billing data using standard protocols.

The mediation device currently creates two output streams: one for billing systems and one for the interconnect system.

Service-desk solution
Ericsson's service-desk solution provides excellent customer-care functionality and keeps track of all customer-related logistics. The system provides an interface for integrating billing systems and enables operators

- to register new customers;
- to register and activate new products and services;
- to manage work orders;
- to manage complaints and faults.

The service-desk solution provides a gateway for existing corporate and external systems using a single view of customer data. It also presents comprehensive, on-line management information on all aspects of calls and their associated work flow.

Billing system
Ericsson's billing system is a convergence billing system that can handle multiple services and produce a consolidated bill for them. The system supports comprehensive rating and discount possibilities with granularity at the subscriber level. A very flexible customer hierarchy accommodates complex company structures, allowing invoices to be produced for multiple levels in the structure and discounts to be collected at selected levels; for instance, at the corporate level. The billing system also has a very flexible tariff structure that allows flat or volume-based discounts to be applied.

The customer base can be split into several billing groups with different billing criteria; for example, due date, interval and customer category. The billing system can also price calls in near real-time, which means that on-demand bills can be created for immediate payment, thus making it possible to produce and print invoices within minutes.

The system accommodates multiple-payment methods; for example, payment records can be collected on-line from banks or giro. The system contains an accounts-receivable function and has a flexible interface to the general ledger for effectively coding and entering financial transactions.

Inter-operator accounting system
Inter-operator accounting is a function that is becoming more and more important as calls are passed between operators before reaching their final destinations. Each operator that handles a call is entitled to receive its share of the call tariff. For this reason, operators enter into agreements with one another. Ericsson's inter-operator accounting system (IOA) collects data on inter-operator calls that are directed to it via the mediation device. It also keeps track of accounting between various operators, allowing settlement according to local and international (ITU-T) rules.

Network operation and maintenance
Good network operations support gives operators an integrated view of the entire network and enables them to assess problems and promptly implement corrective measures when network problems occur. It also provides customer-care staff with accurate information on how problems will affect customer services. Thus, in a competitive

environment, the role of the end-to-end service-assurance process takes on greater dimensions: in addition to serving as a tool for reducing operational costs, efficient network surveillance also has a direct influence on operational income by reducing churn and increasing service availability.

Ericsson's network-operation solution consists of
- a fault manager;
- a performance manager;
- a trouble manager.

These *best-of-breed* applications provide multivendor support; that is, they are not designed for Ericsson network elements alone, but also work with other manufacturers' network elements. The applications have been integrated into a total network-surveillance solution that supports operator processes and the workflow associated with reactive service-assurance tasks. Proactive service-assurance support can also be provided through thresholding and trend supervision, which is built into the performance manager.

In order to satisfy most common customer needs, the network-operation solution has been pre-integrated in a product-development project; for example,
- access modules have been added for receiving alarms from most common network-element managers;
- fault-manager-to-problem-manager integration has been pre-packaged into the standard solution.

Although it has been pre-integrated, the standard solution preserves openness in different applications. The solution also contains several integration reference points (IRP), which are technical enablers that allow other applications—for example, an additional trouble manager—to be integrated into the solution.

The integrated fault manager gives a network-wide view of switching, transport, fixed-access and mobile-access networks. The basic functionality in the fault manager offers a graphical or alphanumeric alarm view. It also enables operators to view the network in domains according to alarm severity and to zoom in on details. The fault manager offers support for building high-performance, distributed solutions.

The fault manager—which supports most common alarm-forwarding protocols, such as ASCII-based, CMIP and SNMP protocols—can retrieve and display network alarms and performance alarms triggered by thresholds and customer-complaint reports. Alarms are forwarded via network-element managers or directly from network elements.

The performance manager provides network-wide reports to users in the operator organization. These include predefined reports for management, planning and operation as well as *ad hoc* reports. Proactive performance monitoring is also supported, alerting network surveillance operators when statistical parameters exceed their thresholds.

The trouble manager distributes and monitors the status of repair assignments inside operation and field maintenance. If a problem is not resolved within the allocated time frame, the trouble manager automatically elevates its status. Service that has an adverse effect is reported to customer care through an information gateway. Integration of the fault manager, trouble manager and customer-care system supports the workflow of operator organizations.

Ericsson's network-operation and maintenance solution supports integration reference points for
- subscribing to network-event reports and receiving customer-complaint reports—these are typically used for integrating a customer-care system;
- the trouble-ticketing interface;
- the performance-monitor interface;
- alarms.

Conclusion

Systems and methods for managing networks, services and customers play a crucial role. Efficient implementation of a telecom-management service infrastructure helps reduce costs, improve quality and speed up delivery of services. For new operators, this means short time to business and rapid payback on investments. For established operators, it means optimizing network operation and streamlining the organization.

Ericsson's commitment to telecom management is based on a common application framework that comprises a comprehensive portfolio of products and services within six management-application areas. It is built out of *best-of-breed*, platform-independent components from independent software vendors and integrates solutions that emphasize scalability and application interoperability between the different areas of telecom management.

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ISSN 0014-0171
Ljungföretagen, Örebro 1998