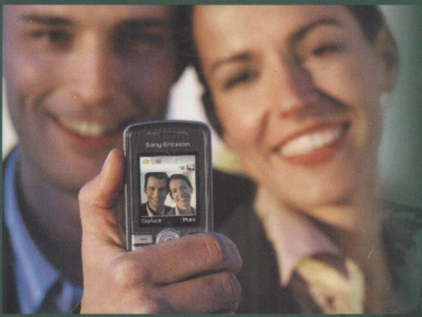

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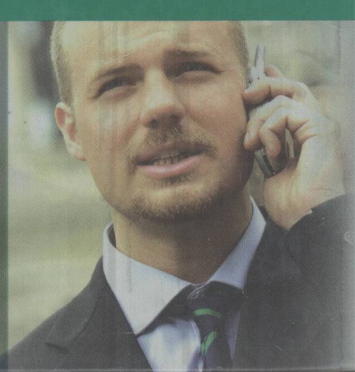

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IMS Multimedia Telephony over Cellular Systems

Preface

This preface is somewhat different from prefaces found in similar books because it does not focus so much on the content of the book. We have instead chosen to write a few words about our own experiences from working with telephony services over Internet Protocol (IP). Here are our stories.

Shyam Chakraborty

In my childhood, black ebonite telephones were a rare commodity and a status symbol. When I made my first telephone call, after a lot of tries and shouting hellos, I could hear a metallic voice through sharp hissing and 'click/clack' sounds. My father told me that it was an art to converse over the telephone, and that it may even be possible to recognize a few voices with sufficient practice. Telephony as an art and as a technology fascinated me. Over the years, I could manage to call effortlessly and talk and chat for hours. And not only identify voices clearly ... it has even been possible to understand emotions over the telephone.

During the late 1980s the extensive proliferation of computers fueled the growth of data communications at a fast pace. Though the present prevalence of the Internet was not then fully understood, forecasts were aplenty that market of data communications would exceed that of voice communications by leaps and bounds. I wondered, even if these predictions are valid, would voice communications take a back seat? It did not. The basic need for telephony got tremendous support from cellular systems due to the offered mobility, portability, good voice quality and wide coverage. Mobile telephony has reached the pinnacle of consumer items, with both grace and utility.

The concept of a converged network has been on the drawing board for quite some time. With meticulous provisioning, the packet switched Internet gains an increasingly convincing role for such a converged network. Rather than talking of voice and data networks separately, a broader concept of services with different quality of service requirements has emerged. A few years back, I became curious whether the wireless interface, despite its 'limited bandwidth', would be adequate for providing real-time services in a packet switched mode, given the different aspects – mobility, security and latency issues – to be satisfied. These thoughts were primarily studied in a more academic setup, somewhat different from that of the rest of my co-editors and authors, who have been studying the design of the radio interface and VoIP services in an industrial research environment. The preliminary results showed me that, as the offered bit rates over the radio interface increased, packet switched real-time services would in general be feasible. This, of course, calls for a clever design of the associated protocol stacks. When I joined Ericsson Research and discussed my thoughts with my colleagues here, I had full corroboration from them.

Mobility and portability have provided fertile ground for a number of conversational and interactive services that are provided more flexibly over a packet switched network. These services allow a richer experience for users in communicating with more information and even personal closeness. Surely, not only the networking paradigms are converging, but a convergence of service paradigms also looms large. I hope this will redefine interpersonal communications in the future.

Tomas Frankkila

During my years within the company I have mainly worked with speech coding for Circuit Switched (CS) cellular systems. This work includes fixed-point and DSP implementation, research, verification of speech quality and standardization. I started working with Voice over IP (VoIP) issues during 2001 and have worked with VoIP ever since. During these years, I have had three 'Aha! experiences' and I will try to describe these here.

When I started with VoIP, most people working in this area were focusing on VoIP over the fixed Internet. VoIP over wireless had of course started but it did not really seem to be realistic to deploy it for a few reasons, mainly these:

1. For wireless systems, one cannot waste half of the resources or more on transmitting the IP, UDP and RTP headers. It is possible to reduce the overhead (per frame) by packing several speech frames into each RTP packet. However, due to the tough latency requirements for full-duplex, real-time voice services, this aggregation needs to be limited to two or maybe three frames per packet, which still gives too much overhead. It was quite clear that, for successful VoIP deployment, header compression would be needed.

2. The Packet Switched (PS) radio bearers were far from optimal for VoIP. For both GPRS and UMTS PS bearers, the latencies were too long. Acknowledged Mode (AM) could not be used because of the quite long retransmission time between the mobile terminal and the RNC, which would give very problematic jitter behavior. And Unacknowledged Mode (UM) bearers were either not available or were too limited to take advantage of the flexibility in IP services.

3. VoIP could not use the radio bearers as efficiently as CS because unequal error protection would not be as optimal as for CS. UDPlite was of course available but it was not as optimal as the super-optimized channel coding and interleaving schemes used on CS bearers.

It was obvious that significant improvements were required. There was also ongoing work to solve these issues, but the work was far from completion in most areas.

One of the most important features that would eventually make VoIP over wireless realistic was the ongoing work with header compression and especially with RObust Header Compression (ROHC). The introduction of ROHC made it clear that the overhead due to protocol headers was manageable. Since ROHC also provided good resilience against packet losses, much better than other header compression schemes, it was quite clear that packet loss due to the air interface would not be a big problem. The problems with inefficient and non-flexible radio bearers still remained.

After working with VoIP for a little while, it became clear to me that VoIP over wireless will actually be better than CS voice. My thinking at that time was that the sound quality of the VoIP service will be better because the great flexibility in IP makes it very easy to introduce wideband speech codecs in the systems. With the development and standardization of AMR-WB it also became clear to me that wideband codecs do not need to have a much higher bit rate than the narrowband codecs used in the existing systems. Previous wideband speech codecs had bit rates in the 32–64 kbps range, which is too high to be useful in wireless systems. With AMR-WB however it became obvious that good wideband speech quality could be achieved at about 12–16 kbps. The complexity of the AMR-WB codec was also manageable, making it realistic to implement the codec in mobile phones.

The first 'Aha!'

My first Aha! experience came when I realized that the quality could be improved by combining:

1. the flexibility of IP, which makes it very easy to introduce AMR-WB;
2. AMR-WB, which gives much better quality than narrowband codecs at a bit rate that is not much higher than for the codecs used for CS, i.e. AMR 12.2 kbps;
3. ROHC, which compresses the headers to reasonable sizes.

Even though radio bearers optimized for VoIP were still not available, and even though unequal error protection was not as optimal as for CS, it was clear to me that the users would appreciate the great quality improvements with wideband speech. In fact I believed that the users would like this so much that they would be willing to pay more for the service and this would compensate for the inefficiencies of the existing radio bearers.

During this time, we were also studying time scaling of speech. This worked quite well, at least for moderate amount of scaling. It became clear to me that a reasonable amount of jitter would not be a big problem.

The second 'Aha!'

The second Aha! experience came in 2003–04 when I learned about the ongoing discussions for high-speed channels. At that time, the general thinking in the high-speed field was focusing on data services and it seemed like they thought that there will be two general types of channels:

- One type of channel is optimized for Transmission Control Protocol (TCP) traffic. This channel type would have short Transmission Time Intervals (TTI), short round-trip time (RTT) and fast retransmissions, which would give low packet loss rates.
- The other type of channel would be specially designed for VoIP. The idea was that this is needed because voice has, as it was said to me, constant requirements for bit rate, packet rate, Frame Erasure Rate (FER) and delay. Since it was also realized that voice is one very important service, one will need radio bearers that are optimized for these requirements.

The short round-trip time and the low packet loss rates were needed to make it possible for the TCP rate control to reach data rates up to the several megabits per second. This actually gave tougher latency and packet loss rate requirements for data than for real-time voice services.

When hearing about this, however, I stated that it is not true that voice has constant FER and delay requirements. The reasons why one uses constant requirements in the CS system is more a design choice than an actual speech property. We had been studying different redundancy schemes for a while and it was quite clear that the quality degradation due to packet losses were much worse for some speech frames than for others. Packet losses gave much larger distortions for onset frames and frames with discontinuities than for steady-state frames. This is because the error concealment, which typically uses repeat-and-mute, works much better for steady-state periods than for transitions regions.

Learning about the short end-to-end delays made me realize that the latency problem was going to be solved for data services, and the transport functions that accomplished the low delay could of course be used also for VoIP. One therefore no longer needed the great quality improvement with wideband speech to compensate for long delays. In addition, it seemed realistic that the low packet loss rates could also be achieved for voice.

Improved service quality would, however, still be needed because VoIP still required more resources than CS because of the non-zero header and since unequal error protection was not as optimal for VoIP as for CS, which gave lower capacity than for CS. Another factor that could probably also compensate for the reduced capacity was the fact that all-IP networks are typically less expensive to operate since one only has one network, the PS network, to manage instead of two networks, PS and CS.

These things made me realize, for the second time, that VoIP will be better than CS voice, even with narrowband voice.

The third 'Aha!'

My third Aha! experience came when learning more about the Hybrid Automatic Repeat reQuest (HARQ) performance and when I was involved in discussions and evaluations on the delay scheduler. When using HARQ, the delay scheduler, and a few other improvements, the capacity for VoIP in High Speed Packet Access (HSPA) was significantly increased and VoIP over HSPA now showed at least as good capacity figures as CS.

So now all components were in place for claiming that VoIPoHS will be better than CS. The quality of the sound will be as good as for CS, since the same codec is used. The performance will actually be a little better for most cases since most users will have lower FER than what they would have for CS voice in UTRAN. Using the same codec as in CS also makes it possible to do Tandem-Free Operation (TFO) with CS, which gives great backwards compatibility and maximizes the quality for interworking scenarios. And none of these optimizations reduced the flexibility, which means that it will still be very easy to improve the quality by introducing AMR-WB.

The end-to-end delay is also not going to be a big issue. The requirements for high bit rates for data services means that short delays are required because of the TCP rate control. The delays actually need to be shorter for data services than for voice, if one wants TCP to reach data rates up to several megabits per second. So data services will actually be the driver for shorter delays and it is natural to use the same transport mechanisms also for VoIP. Thereby the delays will be shorter than for CS for most users under most operating conditions. It is only for the very high loads that the users will experience delays that might be a little worse than for CS.

Conclusion

It is my opinion that VoIP over HSPA will be better than CS for the following reasons:

- The sound quality will be at least as good as for CS voice since the same codec is used and most users will have close to zero FER. The sound quality can also be significantly improved by introducing AMR-WB.
- The end-to-end delay will be about the same as or even shorter than for CS voice.
- The capacity will be at least as good as for CS.

These properties are, in my mind, the most important ones that will enable a successful launch of VoIP in HSPA.

It is my hope that this book will show how to do VoIP over HSPA and also that one should expect as good performance as for CS, or even slightly better, regarding both quality and capacity. Maybe the reader will even experience the same 'Aha! experiences' as I have experienced while working with VoIP?

Janne Peisa

Unlike Tomas, I have spent most of my career in telecommunications optimizing the air interface for IP-based applications. While doing so the focus was (almost) always on the applications using TCP. We quickly realized that one of the fundamental problems with the first cellular packet data access systems (especially GPRS) was the round-trip time (which was close to one second), and we became almost obsessed with reducing the air interface round-trip time. This culminated in the work for High Speed Packet Access (which introduced two millisecond transmission time interval) and Long Term Evolution of UTRAN (which will introduce an even shorter TTI).

It never occurred to me that there would be any interest in providing a high capacity voice service over the HSPA channels we had created. The design goal of the HSPA had always been interactive applications, and we explicitly ruled out any conversational services over HSPA. But suddenly this changed. Preliminary analysis showed that it was theoretically possible to reach or exceed the CS capacity for voice service, and I spent a lot of effort trying to understand how this is possible (for curious readers, the reasons are explained in Section 7.3). The outcome was surprising: when designing the HSPA we had accidentally designed an air interface that was capable of supporting voice applications with higher efficiency than the existing CS bearers could.

As soon as I understood that it would be better to provide the voice service with HSPA access, it also become apparent that suddenly we have *both* the flexibility of the IP-based applications, allowing one to quickly introduce new codecs, add new modes of communications, such as video calls or instant messaging, *and* the efficient performance the CS service.

After redesigning the air interface, it was time to redesign the basic telephony application – to replace the voice telephony with Multimedia Telephony.

I hope the reader can appreciate both the flexibility and the efficiency of the IMS Multimedia Telephony.

Per Synnergren

During my rather brief career in telecommunications I have had the pleasure to work with various nodes belonging to almost all the layers in the ISO/OSI reference model. But the common denominator has always been the end goal of realizing working packet switched communication services.

I started out working with speech coding during the early part of this decade. At that time, much work was being performed in my company, in universities and in the industry in general to optimize the operation of the speech codecs and de-jitter buffering algorithms to secure the voice quality for a voice service running over IP and Internet. Soon companies with Internet telephony as their main business sprung up and released products based on some of the ideas developed during this time frame. For some of the companies the timing was excellent and today we see the success of the IP and Internet telephony business. For me as for many others, it was obvious that IP-based telephony was going to be big business and the discussions about fixed–mobile convergence started to gain momentum.

IMS was the new thing everyone talked about! IMS had been specified in 3GPP release 5 and the first releases of the important base specifications were developed during the time period of 2001–02. However, IMS lacked services. IMS was built to be a general service platform that in theory didn't need any standardization of services. The thinking was that services could be developed by third party companies and just implemented on top of IMS using the ISC interface. But it was soon realized that in practice interoperability could only be achieved by standardization of the services. The first service was PoC (Push-to-talk over Cellular). In 2002, many companies in the telecommunication industry struggled. The operators lost money due to expensive 3G license fees and an increased price pressure on mobile phone calls. It was noticed that one operator seemed to handle the 'bad times' better than the rest, at least in the US. It was NEXTEL, and the specific thing with NEXTEL was their offerings to small and medium businesses. They had rugged phones for the blue collar segment, and they had services that no one else could offer. One such service was Push-to-talk, the cellular walkie-talkie with nationwide coverage. The operators and vendors were desperate to find a new blockbuster application that could help turn the tide around. Maybe PoC on IMS was the savior? Soon an industry consortium was formed that contained Ericsson, Nokia, Motorola and Siemens as the leading players. I ended up as one of many people that worked in this industry consortium producing the set of pre-OMA PoC specifications. This was a really fun time and we all had great hope that PoC was going to be the 'smash-hit' that was to promote IMS. During this time and during the time period I followed the PoC work in OMA I had the opportunity to work with and learn a lot about both the IMS control plane and IMS user plane.

PoC was soon surrounded by hype, but commercially it struggled. The reason soon became obvious. In 2003 and 2004 the commercial mobile networks that were deployed were not good enough to handle the real-time packet switched voice the PoC service produced. At this time the deployment of WCDMA had just started and market penetration was low. Thus PoC had to work over GSM/GPRS to be a success. PoC was designed in such a way that it could be used in a GSM/GPRS network even in situations when only one timeslot was assigned to the mobile terminal. At least it should work in theory, or maybe in a well-planned GSM network that was compliant to the latest 3GPP release. However, the GSM packet switched radio bearer suffered from significantly larger overhead than the CS radio bearer (the LLC and SNDCP overhead). Therefore, the coverage radius of the packet switched voice was significantly less than for CS voice. In reality the commercial GSM networks didn't support all standardized features that were beneficial for PoC and most often the cell planning was optimized for the CS voice service, leading to quality issues for PoC. From that experience I got interested in the radio related issues and I started to work with radio access functionality for IP multimedia.

WCDMA High Speed Packet Access (HSPA) is the most promising way forward. It is certainly not impossible to make packet switched voice services work well over GSM/GPRS and EDGE. For instance, the 3GPP work item EDGE continued evolution may secure the performance needed for the packet switched voice service over EDGE. Another alternative is WCDMA using dedicated channels. But neither of the alternatives above has the same potential as WCDMA HSPA to offer a versatile radio bearer that can deliver the service quality, system capacity and flexibility that allow the operator to do IP multimedia service offerings.

In this book we present the Multimedia Telephony communication service being standardized by 3GPP and promote the idea that Multimedia Telephony has the technological potential to beat the legacy CS telephony service when it comes to capacity and quality at least when utilizing the WCDMA HSPA air interface. I sincerely hope that this will be true also in real implementations. Then maybe in 10 years time we may be able to conclude that the introduction of WCDMA HSPA made IMS and its services become a commercial success.

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Glossary

3GPP	3rd Generation Partnership Project. An international forum responsible for standardizing the GSM and UMTS systems
3GPP2	3rd Generation Partnership Project 2
A-BGF	Access Border Gateway Function
AAC	Advanced Audio Coding
ACR	Anonymous Communication Rejection
ACS	Active Codec Set
ADPCM	Adaptive Differential PCM waveform codec
AEC	Acoustic Echo Cancellation
AF	Application Function
AGC	Automatic Gain Control
AKA	Authentication and Key Agreement
AL-SDU	Adaptation Layer SDU
AM	Acknowledged Mode. One of the modes of the UMTS RLC protocol
AMR	Adaptive Multi-Rate. Speech codec used in GSM and UMTS networks
AMR-WB	AMR wideband. 16 kHz speech codec specified for GSM and UMTS networks
APN	Access Point Name
ARQ	Automatic Repeat Request
AS	Application Server
ATM	Asynchronous Transfer Mode
AV	Authentication Vector
AVC	Advanced Video Coding
AVP	Audio-Video Profile. An RTP profile
BGCF	Breakout Gateway Control Function
BICC	Bearer Independent Call Control
BLER	BLock Error Rate
BSC	Base Station Controller
BSS	Base Station Subsystem
BTS	Base Transceiver Station

C/I	Carrier to Interference ratio. Indication of the link quality
CB	Communication Barring
CCDF	Complementary Cumulative Distribution Function
CCPCH	Common Control Physical CHannel
CCS	Composite Character Sequence
CD	Communication Deflection
CDF	Charging Data Function
CDIV	Communication DIVersion
CDMA	Code Division Multiple Access
CDMA2000	A family of third-generation (3G) mobile telecommunications standards that use CDMA specified by 3GPP2
CDR	Charging Data Record
CELP	Codebook Excited Linear Prediction. General terminology for a group of speech codecs
CFB	Communication Forwarding on Busy user
CFNL	Communication Forwarding on Not Logged-in
CFNR	Communication Forwarding on No Reply
CFNRc	Communication Forwarding on Mobile Subscriber Not Reachable
CFU	Communication Forwarding Unconditional
CGF	Charging Gateway Function
CIF	Common Intermediate Format
CMC	Codec Mode Command
CMI	Codec Mode Indication
CMR	Codec Mode Request
CN	Core Network
CONF	CONFerence
CQI	Channel Quality Indicator. Measurement of the downlink channel quality used for HS-DSCH in UMTS
CRC	Cyclic Redundancy Check
CRT	Cathode Ray Tube
CRTP	Compressed RTP
CS	Circuit Switched
CSCF	Call Session Control Function
CSICS	Circuit Switched IMS Combinational Service
CSQ	Circuit Switched Quality
DCCH	Dedicated Control CHannel. Logical channel used in UMTS
DCH	Dedicated CHannel. Transport channel used in UMTS
DCT	Discrete Cosine Transform

DL	DownLink
DNS	Domain Name System
DPCCH	Dedicated Physical Control CHannel
DPCH	Dedicated Physical CHannel
DPDCH	Dedicated Physical Data CHannel
DRX	Discontinuous Reception
DTCH	Dedicated Traffic CHannel. Logical channel used in UMTS
DTX	Discontinuous Transmission
DVD	Digital Versatile Disc or Digital Video Disc
E-AGCH	Absolute Grant CHannel. Control channel used to schedule transmissions on E-DCH
E-DCH	Enhanced Dedicated CHannel. Improved version of the dedicated transport channels in UMTS systems
E-DPCCH	Enhanced Dedicated Physical Control CHannel. Physical channel used to carry control information for E-DCH
E-DPDCH	Enhanced Dedicated Physical Data CHannel. Physical channel used to carry E-DCH
E-HICH	HARQ Indicator CHannel. Control channel used to for E-DCH
E-RGCH	Relative Grant CHannel. Control channel used to schedule transmissions on E-DCH
EC	Echo Cancellation
ECT	Explicit Communication Transfer
ECU	Error Concealment Unit
EDGE	Enhanced Data rates for GSM Evolution. An updated air interface for GPRS
EEP	Equal Error Protection
EFR	Enhanced Full Rate. Terminology commonly used for improved versions of full rate speech codecs
EIA	Electronic Industries Alliance
ENUM	Electronic NUMbering
EQ	Economy Quality
ESP	Encapsulating Security Payload
ETSI	European Telecommunication Standards Institute
EUL	Enhanced UpLink
FACH	Forward Access CHannel
FBC	Flow Based Charging
FEC	Forward Error Correction
FER	Frame Erasure Rate
FMC	Fixed Mobile Convergence
FR	Full Rate. Often used in combination with a speech codec operating on a full rate channel

FTP	File Transfer Protocol
Gb	Interface between GSM/EDGE Radio Access Network and Core Network
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GIF	Graphics Interchange Format
GOB	Group of Blocks
GPRS	General Packet Radio Service
GPS	Global Positioning System
GRX	GPRS Roaming eXchange
GSM	Global System for Mobile communications
GSM-EFR	Enhanced Full Rate speech codec for GSM
GSMA	GSM Association
GTP	GPRS Tunneling Protocol
HARQ	Hybrid Automatic Repeat reQuest
HDTV	High-Definition TV
HLR	Home Location Register
HOLD	Offline Communication Hold
HQ	High Quality
HS-DPCCH	High Speed Dedicated Physical Control CHannel. Special physical control channel used for HSDPA
HS-DSCH	High Speed Downlink Shared CHannel. Transport channel used for HSDPA
HS-PDSCH	High Speed Physical Downlink Shared CHannel. Physical channel used for HSDPA
HS-SCCH	High Speed Shared Control CHannel
HSDPA	High Speed Downlink Packet Access. An improved air interface for UMTS downlink transmission
HSPA	High Speed Packet Access. Enhancement of the UMTS packet acces. HSPA consists of HSDPA and E-DCH
HSS	Home Subscriber Server
HTTP	HyperText Transfer Protocol
I-BGF	Interconnect Border Gateway Function
I-CSCF	Interrogating Call Session Control Function
IBCF	Interconnect Border Control Function
ICB	Incoming Communication Barring
IDEN	Integrated Digital Enhanced Network
IDR	Instantaneous Decoder Refresh
IEC	International Electrotechnical Commission
IETF	Internet Engineering Task Force
IKE	Internet Key Exchange
IM	Instant Messaging

IM-MGW	IP Multimedia Media GateWay function
IMPI	Private User Identity
IMPU	Public User Identity
IMS	IP Multimedia Subsystem
IMS-GWF	IMS GateWay Function
IMSI	International Mobile Subscriber Identity
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IRS	Intermediate Reference System
ISDN	Integrated Service Digital Network
ISIM	IP multimedia Service Identity Module
ISO	International Organization for Standardization
ISP	Internet Service Provider
ISUP	ISDN User Part
ITU	International Telecommunication Union
Iu	Interface between UMTS Radio Access Network and Core Network
Iub	Interface between UMTS RNC and Node B
JFIF	JPEG File Interchange Format
JPEG	Joint Photographic Experts Group
JVT	Joint Video Team
L1	Layer 1
L2	Layer 2
LAN	Local Area Network
LCD	Liquid Crystal Display
LLC	Logical Link Control. Protocol layer used in GPRS
LPC	Linear Predictive Coding
LTE	UTRAN Long Term Evolution
MAC	Medium Access Control
MAC-d	Dedicated MAC entity. Link layer entity used for Medium Access Control in UMTS
MAC-e	Enhanced MAC entity. Link layer entity used for Medium Access Control in UMTS when using E-DCH
MAC-hs	High Speed MAC entity. Link layer entity used for Medium Access Control in UMTS when using HS-DSCH
MGCF	Media Gateway Control Function
MGW	Media GateWay
MIDI	Musical Instrument Digital Interface
ModIRS	Modified IRS
MOS	Mean Opinion Score