Annex to Report No. 20
Report from the UMTS Forum

Technology Enablers

Annex to UMTS Forum Report: IMS Service Vision for 3G Markets

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1 Introduction

This Annex to UMTS Forum Report 20: *IMS Service Vision for 3G Markets* discusses aspects of the technology for the IP Multimedia Subsystem (IMS) from a high level strategic viewpoint. It identifies Technology Enablers necessary for IMS services in general and for the services specifically examined in UMTS Forum Report 20 and correlates them with the 3GPP work-plan. It also identifies potential delays to the feasibility of IMS deployment due to technical limitations and delays in the availability of necessary underlying technology or technical standards. It also identifies and lists issues relating to the technology, which have to be resolved by the industry.

Today, there is no commonly agreed definition of IMS. However, IMS is now understood to be an evolution of Third Generation (3G) mobile technology that brings the ability to deliver IP-based real-time person-to-person multimedia communication (including IP voice). This means that telephony services can now be fully integrated with other information and data services (including non-real-time and person-to-machine). IMS also has the capability for different services and applications to interact and also the capability for the user to very easily set up multiple services in a single session as well as multiple simultaneous synchronised sessions.

In particular, IMS moves real-time services, such as voice telephony, from the Circuit Switched (CS) Domain to the Packet Switched (PS) Domain, supported by Internet Protocols (IP) rather than the GSM MAP protocols which have their origins in ISDN. It means that all services will eventually be delivered via one integrated network rather than two overlaid networks and this is also expected to bring cost savings in equipment, customer care and network management.

There are three major technology components being developed to enable or support this understanding of IMS and which are the subjects of standardisation work, mainly by 3GPP and IETF. These are:

- The “all-IP” core network including the packet (GPRS) core.
- Multimedia call control (3GPP has chosen SIP).
- The “IP-based” Radio Access Network (RAN) – possibly optional.

What each of these entails is outlined in the following sections as well as the implications of not deploying one or more of these components (work-arounds and limitations). An examination of the service scenarios reveals that certain other technology features will also be needed, not necessarily the subject for standardisation, and these are also outlined in this Annex. Specific technology enablers are identified in the text in **bold print**.

IMS is terminology which has been adopted by 3GPP and is applicable to the IP-based core network architecture which supports the UMTS Terrestrial Radio Access Network (UTRAN) in its Wideband CDMA (WCDMA) and Time Division Duplex (TDD) variants as well as the GSM EDGE Radio Access Network (GERAN). IMS could, in principle, be applied to the 2.5G GPRS packet access network, but it is unlikely that GPRS networks will have sufficient capacity to make it worthwhile. 3GPP2 also has an “all-IP” project for its core network and cdma2000 radio access network, which also includes a multimedia call model and IP-based RAN.

2 Evolution from GSM to IMS

Table A1 shows how digital cellular features have evolved from the beginning in GSM, through GPRS and 3G to IMS. It is a simplified guide, showing only the main features and with intermediate releases omitted:

Table A1: Evolution of digital cellular features

	2G (GSM)	2.5G (GPRS)	3G (UMTS)	IMS
Features Which Are New	Circuit Switched Digital Speech Telephony Circuit Switched Data (Dial-Up) IN-based Supplementary Services SMS	Packet Switched Data (Always On) Non-Real-Time Internet Services WAP	Broadband Packet Data Multimedia Messaging Services – MMS (Person-to-Machine) Video Download and Broadcast MMS Instant Messaging XHTML	IP-based Real-Time Person-to-Person Multimedia including Packet Voice (VoIP). IP-based Telephony Services. IP-based Packet Video Telephony. Integrated Real-Time & Non-Real-Time Services Multiple Simultaneous Services & Sessions
Features Which Continue		Circuit Switched Speech Circuit Switched Data (Dial-Up) IN-based Supplementary Services SMS	Packet Switched Data (Always On) Non-Real-Time Internet Services Circuit Switched Speech IN-based Supplementary Services	Packet Switched Data (Always On) Non-Real-Time Internet Services inc. Video MMS Instant Messaging XHTML
Legacy Features Supported			Circuit Switched Data (Dial-Up) SMS WAP	Circuit Switched Speech IN-based Supplementary Services SMS
Obsolete Features				Circuit Switched Data (Dial up) WAP

Source: UMTS Forum

To illustrate how the table works, consider as an example circuit switched data (dial-up access). This was introduced as a new feature in GSM.[1] It is a feature that continues into 2.5G, but packet switched data (GPRS), which is now newly introduced, provides a faster service with the convenience of "always-on" connection. 3G (UMTS), which goes a stage further with broadband packet data, still includes a GSM-based circuit switched domain and so circuit switched data will probably exist as a legacy service. However when IMS is deployed, bringing real-time services such as speech into the packet domain, circuit switched data will really be obsolescent and operators will be considering whether to discontinue it rather than deploy interworking facilities which would enable the service to be replicated in the packet domain.

[1] Actually GSM Phase 2.

Table A1 shows that circuit switched digital speech telephony, which has continued as a service since the start of GSM and all the way through 3G, is replaced in IMS by IP-based packet switched voice telephony. Circuit Switched Speech has to be supported as a legacy service in IMS, because of the large number of conventional terminals (fixed and mobile) which will still be in use. The GSM IN-based Supplementary Services are replaced by IP-based telephony services in IMS which either replicate the old ones, or more likely replace them by ones which will be much richer and very easy to use.

The UMTS Forum identified six service classes in its earlier 3G market forecast studies.[2] These are; Mobile Internet Access, Mobile Intranet/Extranet Access, Customised Infotainment, Multimedia Messaging Service (MMS), Location-based Services and Rich Voice. For brevity, only MMS has been shown in Table A1, however all except Rich Voice would be in the same positions. Rich Voice is, in effect, the enhanced voice service provided by IMS, together with the Video Telephony Service which in the 3GPP specifications is a packet switched domain service.[3] It should be noted that these 3G services (except Rich Voice) can and are being implemented early on GPRS (2.5G). However, because services such as MMS were specified by 3GPP, targeted at UMTS, and because of the limited capacity for mass-market 3G services on GPRS, we have classified them as 3G services. Simple Voice is the term that the UMTS Forum uses to describe basic speech telephony services without the rich features that IMS can bring.

[2] UMTS Forum Reports 9 and 13.

[3] Unlike the initial Japanese 3G FOMA Videophone service which has used the circuit switched domain.

3 What IMS Delivers

IMS provides the capability for person-to-person real-time multimedia, integrated services, customised telephony services and multiple synchronised sessions.

3.1 Person-to-Person Real-Time Multimedia

In most instances "**person-to-person real-time multimedia**" will mean speech telephony. In future, it will also increasingly include video telephony. The move to IMS means that these services will be provided in a single packet based domain based on the Internet Protocols (IP) which are becoming the common standard across the worlds of telecommunications, information technology, entertainment, telemetry and telecommand. It means that in principle, in the future, a mobile videophone user could communicate with a group of people who are using a conventional TV display connected to a cable box or a PC display and broadband modem. All these technologies (mobile, broadcasting and fixed telecommunications) are moving to IP.

3.2 Integration of Services – an Example

The capability to integrate telephony with other information and data services and enable them to interact is best illustrated by considering directories and how calls are set up. Today, making a call to someone involves remembering their number or looking it up in a paper directory and then dialling it manually on the keypad, or alternatively using the phone's own stored directory which would have been laboriously entered manually by the user on some previous occasion. If a directory is not to hand and the contact is not in the phone's directory, if the called party's number has been forgotten or if the number or area code has changed, then the user's only recourse is to use directory enquiry services. The directory enquiry service has to be contacted and the number spoken by them remembered and entered through the keypad, which is not an easy process if the mobile user is on the move or overseas.

With an IMS phone however, the integrated information/data services will enable on-line directories to be provided and because the data part is packet switched "always on", these on-line directories will be as readily available as a directory stored on the phone (off-line). An on-line directory is always up to date, because it is updated automatically by the network when subscriber details change. Furthermore, once the contact has been found in the directory, the user only has to click on the name and then the number (or URL in the case of IMS) will automatically be dialled. Imagine having the worldwide Yellow Pages instantly on tap, with quick and easy hierarchical menu-driven search! For personal contacts, the worldwide phone book will be available albeit a little unwieldy, however advanced interactive web-based search engines will quickly find the contact and again, the user will only have to click for the contact to be completed.

The **Presence** service will further enhance the experience by informing the caller if his contact is available at that instant and if not, offering alternative means of contact (SMS, email, voice mail, etc.) which be just a click away. Presence will particularly enhance services for mobile users and mobile contacts, especially when it is combined with **Location Services** (LCS).

Of course 2.5G and 3G (Release 99 and Release 4) have broadband "always-on" packet data with simultaneous speech and data capability and so can provide access to on-line directories. However without IMS, the directory page can only display the contact number, which still has to be entered digit by digit on the phone's keypad in order to complete the call. By contrast, with IMS, the user only has to click on the contact – he need never know the number. It is this **integration of services** and the **interaction of services** between the information data service (the directory) and the telephony service (the call control) which really differentiates IMS.

3.3 Customised Telephony Services

In the case where the contact is an organisation rather than an individual, the organisation itself will be able to customise the appearance of the information provided and hence the ways that calls are made to it. Layered information can be provided, just as for any web site. The top layer might provide the initial contact point for the organisation, selecting this would take the user down a level and offer him contacts with the CEO's office, the CFO's office or the CTO's office. Alternatively he can go down a level where he would be offered contact with Customer Services or the employee directory. The point is that this structure and the ensuing call options would be individually tailored to the structure of each organisation.

This is an example of **customisation by the service provider**, which is another capability that IMS provides, rather than customisation by the phone manufacturer, which is the case for 2G and basic 3G telephony.

In the example above, a visual display and pointing device has been implied for the directory. However, it could equally be a voice browser which accesses the on-line directory and in this case the audio information and voice commands in response would fully control the call set up, without any need for the user to worry about a phone number. This is an example of **customised integration at the application level**, which is provided by IMS.

3.4 Multiple Synchronised Sessions

Another feature of IMS is the ability to set up **multiple synchronised sessions**, which include any mix of media (e.g. voice, video, text) or mix of real-time communications with non-real-time information, person-to-person, person-to-machine or machine-to-machine. Calls within calls (such as "private chat") are included in this.

In the example above, having found his contact through the personal on-line directory or search engine and set up the call, the two parties can browse through the yellow pages together (synchronised and controlled from either end), to find a supplier and a product of mutual interest. They can then call the supplier and speak to him together, occasionally breaking off to discuss their opinions together privately. The supplier can send them pictures of the product and then when they have decided, they can order, confirming the credit card transaction with the bank.

However, without IMS, each of the sessions[4] would all be separate actions involving setting up the terminal. Browsing yellow pages could only be done independently by the two parties while the "private chat" would involve the fiddly use of the call hold function. By contrast, with IMS, it is all one seamless point and click process shared by the two parties together.

An issue arising from the above scenario is that **IPv6** is an essential enabler to support the resulting need for numbering and addressing. The pace of development and deployment of IPv6 is cause for concern for some.

3.5 Impact of IMS on Other 3G Services

This study has concluded that an IMS capability is not essential for near real-time services such as Instant Messaging or Multimedia Messaging Services (MMS). However, the improved Quality of Service possible within IMS (see Section 4.2) could undoubtedly enhance the users' experience in many cases. Also with IMS, Instant Messaging or MMS features can be integrated with speech telephony to provide new and innovative combined services.

[4] Accessing the directory, setting up the voice call, browsing the Yellow Pages, setting up the call to the supplier, sending the picture of the product and making the credit card purchase.

UMTS Forum Report 20 has examples with use case scenarios of services that take advantage of the features of IMS described above. However it is necessary to consider how each of the features is impacted by the technology, especially the following three major technology components already introduced in Section 1:

- The “all-IP” core network.
- Multimedia call control.
- The “IP-based” Radio Access Network.

Such considerations are vital in order to understand the impacts of any delays in standardisation or development timescales as well as the scale and the cost. The effect of partially deploying the technology or deploying work-arounds can also be deduced and are discussed further in later sections of this Annex.

4 Comparison With Fixed Network Internets and Internet Telephony

Internet-based voice telephony is already being introduced in fixed networks [1]. In the fixed network world it is generally known as Voice over IP (VoIP). Perhaps because the mobile world has recognised that other forms of real-time person-to-person media can be used (particularly video telephony and gaming), it has adopted the wider term IP Multimedia Subsystem (IMS). However, from the users' perspective VoIP and IMS will deliver broadly the same rich services. Much of the motivation for IMS has come from the fixed networks and it is worthwhile looking at the similarities and differences in order to make a start in understanding where new technology is needed.

4.1 Network Architecture

Common to both VoIP and IMS is the convergence of the previous circuit switched TDMA voice network and the packet switched data network into a single packet-based network carrying voice, data and multimedia traffic. Also common to both is the **separation of the control and user planes**, meaning that the control and service mechanisms are implemented in separate control servers, which are physically separate from the core transport infrastructure. This is the key to the ease of introducing new services in IMS. Code simply has to be written in a common high level language for a control server which can be connected anywhere in the network rather than code for an ISDN switch which is generally special and proprietary and is located according to the network traffic.

4.2 Quality of Service (QoS)

Existing Internet networks do not meet quality requirements for the low delay and jitter needed for real-time services such as digital speech telephony. Where VoIP is being introduced by fixed network operators with the intention of achieving PSTN quality it is being done through the deployment of new self-contained autonomous global Internet networks designed to meet QoS requirements for speech. These autonomous networks allow for the provision of adequate bandwidth to avoid congestion (thus delay) combined with control over access to bandwidth. The two key technologies that enable the QoS requirements to be met are the **Resource Reservation Protocol** (RSVP) and the **Multi-Protocol Label Switching Protocol** (MPLS). RSVP provides a mechanism for setting up paths across a network that can meet the specific QoS (bandwidth and delay) requirements for a given type of traffic. MPLS speeds up routing decisions for individual packets both at individual routers and also at the network level by setting up a complete route across the network (the Labelled Switched Path – LSP). Thus MPLS deals with the jitter aspect of QoS by reducing processing delay and congestion. An LSP decision can be influenced by RSVP and so the two technologies work together. What it means is that upgraded routers are needed in the VoIP-supporting Internet – MPLS enabled and with RSVP capability.

4.3 Interworking with Conventional Networks

In order that VoIP users can communicate with conventional PSTN users, **gateways** are provided between the two domains while **softswitches** (special servers in the VoIP network) replicate the ISDN switching to enable interworking to take place. In particular, **interworking between E.164 numbering schemes and IP addresses** is provided for.

4.4 Special Requirements for Mobility

Thus far the VoIP and IMS network requirements are broadly similar. However the requirements for wireless and mobility add significantly to the considerations in the IMS case. With a tethered subscriber and the inherent security of the wired network it is much simpler for the fixed network provider to re-engineer **billing**, **security**, **privacy**, **authentication** and **registration** for VoIP. There is no need to implement features such as **roaming**, **handover** and the **Virtual Home Environment** (at least until terminal mobility

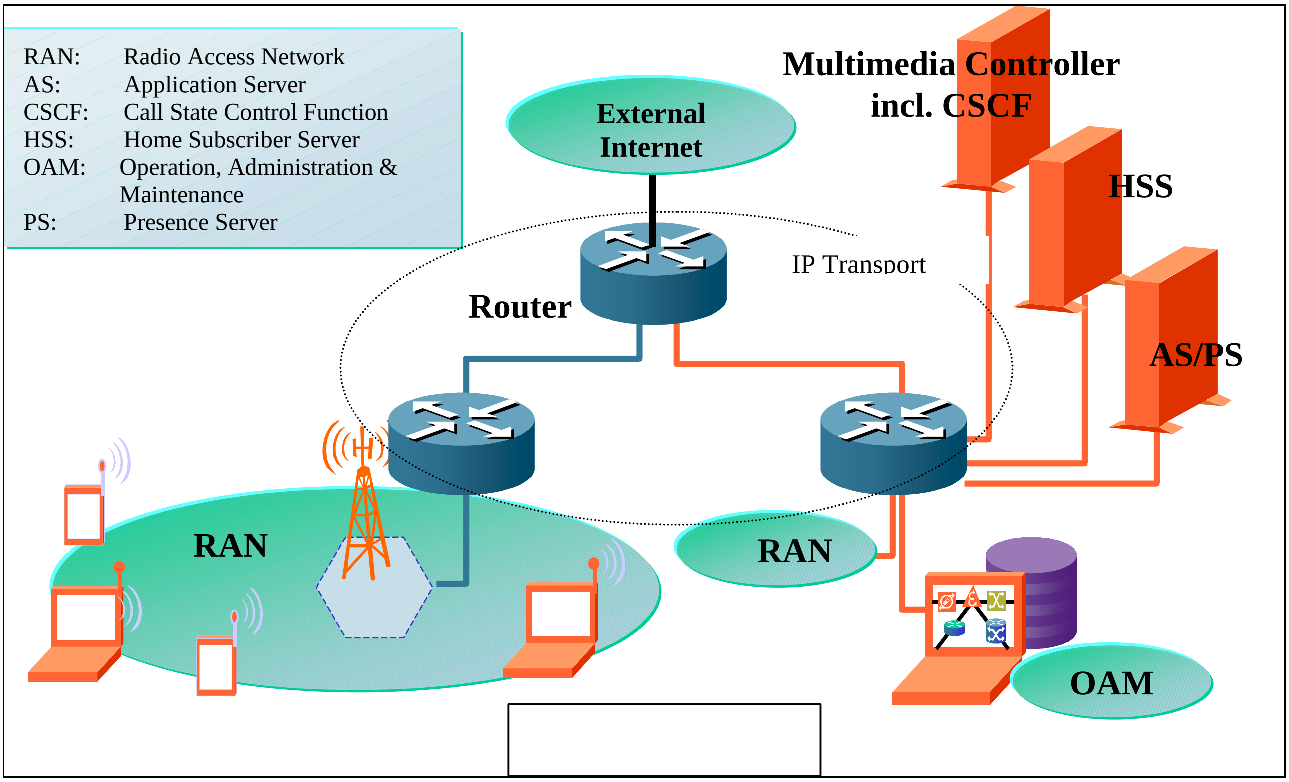
comes to the fixed network). All these aspects are or will be major work items for standardisation in 3GPP or IETF for IMS.

Significant improvements have been made in modem technology over the past few years, and the data rates that can be sent over ordinary telephone wires (the “last mile” to the subscriber) have increased dramatically. ADSL and cable modems achieve 384 kbit/s and higher, while the latest PC modems connect at 115 kbit/s via an ordinary telephone socket. Given that good quality digital speech only requires 13 kbit/s, it can be seen that the fixed access network now has massive extra capacity. This is important, because VoIP has a heavy overhead, not necessarily for any rich content but even for basic speech, due to the high payload on IP packets for routing and control data. The over-capacity on wired networks easily provides for this overhead and VoIP terminals are being installed on existing infrastructure and subscriber lines [2]. The same situation is not true for cellular mobile where efficiency for the transport of basic speech over the air interface is critically important, affecting spectrum efficiency, capacity, revenues and infrastructure costs. This issue is discussed further in the section on the Radio Access Network (Section 7).

5 The "All-IP" Core Network

The "All-IP" Core Network was the start of IMS. It was brought into 3GPP, largely through the activities of 3G.IP and MWIF as an architecture concept before service principles had been articulated. A key driver was commonality with the fixed networks, which as we have seen are moving to an IP basis. It was also thought that an IP-based core network would have lower life-cycle costs. A simplified overview of the IMS core network is shown in Figure A1:

Figure A1: IMS core network



Source: Siemens

5.1 IMS Core Network Platforms

Development of the IMS core network itself involves the provision of **seven new platforms**:

The **Call State Control Function** (CSCF) manages the SIP session establishment and call control and forms the link to the **SIP Application Server** (AS). The **Policy Control Function** (PCF – not shown) manages the Quality of Service policy. The **Multimedia Resource Function** (MRF – not shown) controls the multi-party conferencing features of SIP. A **Media Gateway Control Function** (MGCF – not shown) together with a **Transport Signalling Gateway** (TSGW – not shown) perform a similar function in the mobile domain to the softswitch in a VoIP network, i.e. they enable interworking with ISDN-based circuit switched networks including interworking between IP addresses and E.164 numbering schemes. The **Home Subscriber Server** (HSS) is the packet (GPRS) equivalent of the HLR in GSM and carries through from 2.5G/3G into IMS. In addition, an **MSC server** could be added, which is another form of softswitch enabling this same all-IP packet switched core network to deliver GSM circuit switched services via a 2G or 3G RAN to conventional mobile terminals. The **Wideband AMR speech codec** is mandatory for IMS. There is one codec per trunked voice channel and so the investment will be quite significant. The IMS network and SIP call control place new requirements for the **Telecom Management** system (shown as OAM – Operations Administration & Maintenance in Figure A1) including

Charging applications. **Terminal Management** is also required as part of the OAM, in order to provide customer care for the User Equipment (UE) which will have the capability for downloadable applications.

5.2 Edge of Network Control

The control and application servers are placed at "the edge of the network" and this results in the separation of control and user planes. What this means is that the call processing has been physically separated from the switch platforms (routers in packet technology) that remain in the transmission path, but are relatively unintelligent compared to a conventional ISDN-based switch. The IMS processors are built on standard platforms and run an industry standard operating system and software language. It is this factor plus the concentration of the processing in separate centralised platforms rather than distributing it across many specialised switch platforms in the network that makes it much easier for innovative services to be developed by entrepreneurs.

The interactions of the separated call control processor in the network and the terminal result in header information being added to the packets at source together with the user data. The header information is read by the routers and enables them to route the packets to their destination and to respond to the appropriate QoS strategy. The separation of control and user plane processes should not be confused with "in-band" and "out-of-band" signalling used previously in ISDN.

The fact that the network transmission backbone is relatively unintelligent does not mean that the network operator becomes just a bit pipe provider. The call control and application servers that are the heart of the innovative and value added services are still part of his network. It is true, however that in principle, the technology enables a would-be third party service provider to connect his own platforms to the network and provide his own innovative services. Mobile operators could prevent this through the use of security mechanisms but in any case, they have a head start in understanding the mobile services business. The mobile industry therefore needs to decide to what extent the IMS telecommunications model in this new era will follow the Internet model by allowing services to be provided externally.

5.3 Standardisation and Deployment of the IP Core Network

The core network standardisation is due for completion by 3GPP in Release 5, well ahead of the completion of the other components. Deployment of the IMS core network alone will not provide the capability for rich IMS services described above. It would only be capable of replicating the existing 2G/2.5G services. Nevertheless, there are arguments for just deploying the core network part. Obviously where the mobile network is closely connected with an IP-based fixed network, there are arguments in favour of commonality. Another consideration is the deployment of 3G networks that are "IMS ready" for voice and real-time services rather than deploying the circuit switched part of the 3G network, which will have to be replaced. Vendors have offered all-IP core networks with MSC servers replacing the traditional MSCs, but so far, this approach has not been adopted.

There is an alternative method of providing IMS services based on the deployment of only the IMS core network. It can be achieved by placing a **SIP Proxy** in the core network. This approach could enable the provision of IMS services earlier and possibly at lower initial cost, however there are significant issues and disadvantages that have to be considered. The subject is covered further in the section on Multimedia Call Control (Section 6).

5.4 The Core Network outside the 3GPP PLMN

Assuming that the intention is to provide the full IMS capability, including multimedia call control, then there are still deployment issues regarding the core network which have to be solved. The IMS core network architecture provides for the necessary QoS capability to support real-time services – two-way speech and video telephony. However the delay, jitter and packet loss rate must be maintained end-to-end. From looking at architecture diagrams, one could be forgiven for thinking that the IMS core network exists just within the

confines of the PLMN. It is often shown connected externally to an “Internet cloud”. However from the foregoing it is clear that “any old Internet” will not do. Any intervening IP network must also support the QoS requirements (in effect be VoIP enabled) incorporating adequate bandwidth, RSVP and MPLS. An issue for operators provisioning for links between PLMNs is that global VoIP-capable Internets are few and far between and being rolled out relatively slowly over the next 10 years [1].

An alternative, which might be proposed, could be to simply provide for the inter-PLMN links with dedicated fixed capacity. However, since such non-VoIP links would have no knowledge of the QoS requirements of the traffic, they would have to be provisioned for the worst case. Non-urgent packets (e.g. for email) would have to be transmitted with the same delay and jitter required for real-time packet speech. This would result in massive over-provisioning of expensive capacity. By contrast, with IP networks built for the purpose, VoIP network operators benefit from savings on their fixed high capacity links due to the intermittent nature of speech packets.

An operator may also need to look at components within his own packet network. The GPRS core may need upgrading with MPLS-enabled servers and increased capacity in the transmission backbone.

5.5 Roll-out Considerations

Unlike most of the 3G services previously analysed by the UMTS Forum, which were information or entertainment services (server-to-client), IMS provides mainly for person-to-person services. Taking as an example, voice calls in today’s networks (which, of course are person-to-person calls), most calls originated in one network terminate in another, often competing network. This is particularly true of mobile networks where there tends to be more choice and competition and because many mobile calls are to subscribers in fixed or private networks. Operators deploying 2.5G/3G services can do so at their own pace according to the needs and preferences of their own customers, the drive for competition and their capacity for investment. What they do is little affected by what is done in other competing networks. By contrast however, for IMS person-to-person services, there is much more dependency. If a group of callers are to enjoy an IMS session together, then all their networks must support IMS. Although MSC servers and softswitches can provide for basic calls between IMS networks and legacy networks, the potential richness of IMS services is lost. There is therefore a much greater need for a **concerted roll-out** of IMS core networks by the industry, otherwise IMS subscribers will not get the improved services even if their own network provider has invested in the technology. It could lead to another case of disappointment by cellular subscribers if expectations are not met. The need for a more harmonised roll-out takes the industry back to the earlier days of GSM deployment when this was also the case and the subject is dealt with for the other components of IMS in later sections.

At a workshop held in Toronto April 3-4, 2002, 3GPP and 3GPP2 agreed to recommend the implementation of a common network architecture for IMS. This is particularly significant because it means that the introduction of IMS would, for the first time, result in a harmonised standard architecture for cellular worldwide.[5]

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[5] The meeting agreed that both 3GPP and 3GPP2 should adopt a single IMS reference model and work to ensure interoperability and application-level intersystem IMS roaming.

6 Multimedia Call Model

In order to provide for the interaction, integration and customisation of services as well as for multiple sessions involving mixed media and real-time combined with non-real-time services, as described above, a multimedia call model is required instead of the ISDN C7-based MAP call model used for 2G. After some deliberation, 3GPP chose the Session Initiation Protocol (SIP), which is being standardised by IETF. The alternative was H.323, which is an ITU specification with more of a traditional telephony background. It is interesting to note that SIP has been incorporated as part of Microsoft Windows XP. H.323 was incorporated in earlier versions of Windows and is used as the basis of Microsoft NetMeeting.

Users of NetMeeting will therefore have had some experience of IMS type services, especially shared whiteboards and presentations, which involve the set up of multiple media sessions. It has to be said however, that the current early versions of this multimedia communications service are not particularly user friendly and the speech telephony feature, which is incorporated, is rarely used because of the poor quality often experienced on non-VoIP engineered Internets. Users currently tend to prefer to use a separate conference phone. No doubt the situation will improve rapidly. Nevertheless the fact that SIP can and is being integrated into PC operating systems illustrates how this form of call control can be accessed by third party applications through a standard API. More importantly it can be accessed remotely by a web application, enabling service providers to set up customised services. The operation of embedded SIP can be thought of as similar to the operation of other multimedia components, which can be accessed by third party applications and web pages. A simple example is Media Player, which allows applications such as PowerPoint and web pages to display video as if they were on the page. In the case of SIP it would be the setting up of calls and multimedia sessions within other applications including web.

6.1 SIP Standardisation and Deployment

SIP is therefore the cornerstone to the provision of the rich IMS services described throughout UMTS Forum Report 20 and this Annex. Its standardisation however, is somewhat behind the standardisation of the core network and all features for it will not be completed by Release 5. An issue is the completion of **SIP extensions** requested by 3GPP from IETF for mobility roaming and security.

From a deployment point of view, new terminals supporting SIP are required. Call control in the network is provided by a **SIP server**, which is an “edge of the network” platform. In principle, multimedia calls could be completed from a SIP network to a subscriber who is in a network with an IMS core but in which a SIP server has yet to be deployed. However, it is extremely unlikely that an operator would deploy an IMS core network without the intention of providing a SIP server to enable his customers to enjoy reciprocal services. Therefore, as with the IMS core network, there is a good argument for a co-ordinated deployment approach for SIP. The same could be said for a range of key basic SIP-based IMS services so that there is a good degree of common availability provided for users in different networks (who, remember are in close communication person-to-person with IMS) in order to avoid disappointment amongst groups of them.

6.2 Development Toolkits for SIP-based Services

SIP services, designed for implementation on a network server rather than in an ISDN switch, are inherently much easier to develop because **standard development tools** will be used to develop IMS services in much the same way that innovative Internet services have been produced by entrepreneurs. However, for mobile services there is added complexity due to the need to take into account mobility management, security, privacy, authentication, registration, accounting and billing (including third party) as well as service management and integration. New **toolkits** need to be developed and be made widely available to ensure that innovative services will be available. The European Commission IST Directorate has recognised the need for this and it has initiated a number of fast track R&D programmes in support.

SIP provides for true end-to-end IP applications, thus enabling the vision of rapidly created customised multimedia communication services. It also means that IP is transported end-to-end (terminal-to-terminal and terminal-to-servers) and so the separation of control and user planes is maintained throughout. In addition to new terminals incorporating the **SIP protocol stack**, there are issues for the radio access network because of the overheads for IP transport described in Section 7.

6.3 Alternatives to End-to-End IP

It has been suggested that as an alternative to providing SIP end-to-end, a **SIP Proxy** could be deployed as a network server. Some specialist suppliers appear to be offering this as an option. It would enable existing 3G terminals to access SIP services and would remove the need for the radio access network to transport IP with its associated overheads, particularly over the air. It has also been suggested that **Java-based phones**, which will be deployed for 3G, would be able to replicate the look and feel of true customised SIP services by means of **downloadable software**. However there are a number of issues to be considered. It means that there would be a special **SIP Proxy-to-Terminal interface**, which would be unique to the mobile industry. The layers below the application would be unique, not just to mobile but probably also specific to each terminal type. The mobile industry would be faced with maintaining this interface into the future and as a legacy even if the full terminal-based SIP solution was adopted at a later date. The downloadable software would also need to be maintained, possibly needing to be revised for each new class of service and type of terminal. There is also the possibility that some non-SIP terminals would not replicate SIP services well, despite the downloadable software and this could lead to customer disappointment. It means that the mobile industry would diverge from the fixed telecommunications industry where SIP-based terminals can be introduced more easily into the existing infrastructure. Thus the industry needs to consider very carefully whether to go down this route.

7 Radio Access Network

7.1 IP Transport in the RAN

The packet transport in the fixed backbone of the 3G Radio Access Network (RAN) must meet the QoS requirements for real-time services. Transport can be provided by conventional means such as ATM fixed links with some over-provisioning for peak data loads. It is not therefore a major issue. However, **IP transport** is relatively straightforward and follows the same principles for the core network and is a current work Item in 3GPP. IP-based transport for the RAN backhaul is expected to provide some cost benefits when it can be deployed.

7.2 IP over the Air Interface

Although the 3G RAN has the additional bandwidth to make multimedia IP over the air feasible, the overhead incurred when an IP bearer is used to transport basic speech over the air interface reduces **spectrum efficiency**. It is a critically important measure affecting capacity, revenues and infrastructure costs. It was one of the key market drivers from the very start of GSM development and has remained a key factor in promoting the developments of system enhancements such as advanced speech codecs, frequency hopping, microcells and picocells. It is a major competitive factor in the considerations of CDMA *vs*. TDMA. Enhancements to the standards are being developed to reduce the impact under the heading "**IP-based RAN**" or sometimes just "All-IP". These include **Robust Header Compression** (ROHC), and **SIP Signalling Compression**, which will be developed for 3GPP by IETF. However, this work is apparently behind the rest of the standardisation and even with ROHC, spectrum efficiency may still be impaired compared to GSM and 3G. There are issues remaining to be solved for ROHC including IP security and handover.

We found that opinions within the industry differed considerably on whether improvements to the UTRAN are essential for IMS. One of the problems is that the metrics for the effects on spectrum efficiency, with or without ROHC have not been widely published or discussed. Therefore the industry has not been able to make a decision on whether to ignore the problem, accept a temporary reduction until ROHC is available or whether to seek new ways of overcoming the problem through new research. One source quoted an improvement of 14% on the uplink and 6% on the downlink when header compression was applied to an IP bearer carrying IPv4. However, the improvement was said to be 75% when IPv6 is carried. The improvements will almost certainly be essential for GERAN because in most cases the 3G spectrum available for those networks (principally in the USA) is less and hence the capacity is more critical.

The mobile industry needs to come to a decision on whether the IP-based RAN should be deployed from the start, whether it should be deployed later or whether it should remain an option. The cost and complexity of upgrading 3G RANs to "All-IP" will undoubtedly be a factor (and one that seems to differ between suppliers). The industry may also need to decide whether to mandate such improvements in the IMS terminals, so that the spectrum efficiency improvements will be fully realised when the improvements to the infrastructure have been made.

7.3 Other Considerations for the RAN

Other capacity improvements are also under way in 3GPP including **High Speed Downlink Packet Access** (HSDPA) and its uplink equivalent. However, these capacity enhancements do not directly tackle the problem of spectrum inefficiency and they will not work under all circumstances. It also needs to be determined whether any of the solutions can be retrofitted to existing 3G RANs.

There is much discussion in 3GPP over which 2G speech services should be replicated for IMS. Particular examples include free emergency calls (which have complexities of implementation in the packet domain), the IN-based **supplementary services** and CAMEL

(which is an important enabler for pre-pay phones). Fortunately at least, since IMS services are implemented in edge-of-network servers, the re-development will be quicker and easier than for the originals. However, the technology for providing such services together with the basic call control itself will be very different between IMS (SIP) and 2G/3G (MAP). The deployment of both MSC servers and SIP servers would enable calls to be set up either as legacy calls or IMS calls at the start of the session (depending on network and terminal capabilities for all the participants). But providing for **in-call handover between IMS and CS domains** would probably be extremely complex. It would be impossible according to one notable 3GPP expert in one of the presentations referenced. Operators therefore need to plan to deploy IMS-capable access throughout their whole network coverage so that inter-domain handover is not necessary.

7.4 Access Independence

One of the key advantages for IMS quoted by many experts is that it makes **access independence** much easier to achieve. Access independence means that IMS calls can be delivered to suit the situation of the terminal via alternative access networks such as alternative IMT-2000 RAN technologies, wireless LAN, fixed or private networks. However, using IMS to provide for access independence tends to drive the solution towards the so-called **loosely coupled model**. This may not necessarily be the best model with regard to operators' business interests because it enables access to be provided via privately-owned or competitor-owned LANs very easily. 3GPP has only just started looking at this topic under the project heading **Wireless LANs**.

8 Implications

The implementation of IMS has implications beyond the introduction of new technology into the core network and the RAN. Some of the key issues are addressed in this section.

8.1 Terminals

It has already been pointed out that terminals will need to incorporate new technology to support the multimedia call model. The richness of the services made possible by IMS means that high standards for Man Machine Interfaces (MMIs) will be necessary to take full advantage of them. This would include **high resolution displays**, **innovative audio**, **compact browsers** and easy to use **pointing devices** as well as **sufficient memory** to run the applications well and incorporate facilities for downloadable client software. A number of people in the industry have suggested that powerful browsers will not be practical for small handheld devices for some years. The impact of this needs to be understood. MMI and performance aspects are mainly proprietary to terminal manufacturers. However, the mobile industry might consider it advisable to specify certain minimum terminal capabilities for IMS to ensure full enjoyment of the services by users. There is a precedent for this in the IT industry where software vendors specify minimum requirements on PCs and peripherals to ensure good results from their products.

There has been a tendency to consider that 3G information services will be used mainly by nomadic rather than fully mobile subscribers because the information provided can be more easily digested. Person-to-person IMS services, especially speech telephony, will often be required by mobile users. If these mobile users are drivers they will need hands-free, eyes-off operation. This brings to the fore the need for **voice browsers** and **voice recognition** so that the richness of the integrated services can still be provided. Both these topics are in the standards domain (WAP forum and 3GPP respectively) although for completion beyond Release 5.

The SIP call model and the use of IP-based security mechanisms requires a new type of Subscriber Identity Module (SIM) application, known as the IMS SIM or ISIM.

8.2 Network Optimisation

Although not strictly a technology enabler, the technical resources needed to tune networks in order to meet commercial quality standards should not be underestimated. Packet switched mobile speech telephony is a completely new untested concept and it is likely that the process of testing, fault finding, upgrading and re-designing networks and software will be as intensive as for previous ground-breaking technologies. Results from GPRS networks show that while the delay floor for packets is approaching the 50 ms needed for IMS, there are regular and significant peaks of up to 3 seconds. It is thought that this might be due to ping-pong handovers or peaks in offered data traffic. Significant numbers of packets are being “lost” and while this can be accommodated for non-real-time data services through the use of ARQ, it is unacceptable for speech. To make matters worse, these results are occurring on networks that are virtually unloaded. While the performance for packet data will undoubtedly improve on 2.5G (Release 99) and 3G (Release 4) networks, it is unlikely to reach the performance needed for IMS because there is no need to support real-time services in them.

At the introduction of both GSM, GPRS and now 3G, the optimisation of networks in the field, together with back office support for software re-development and changes to the standards have involved teams running into hundreds from each supplier working over many months. Testing must be done end-to-end and network-to-network and there may be a need for **special test equipment**.

8.3 Deployment Aspects

Analysis of each of the technology components has shown a case for a more concerted approach to standardisation, development and roll-out than for 3G. Another argument in

favour of this comes from the fact that IMS is mainly for person-to-person services and that requires new technology that is not directly compatible with the old. In cases like IMS where a user can only utilise a service together with other similarly equipped users, the take-up of the service will be geometric rather than linear.[6] To optimise growth in a geometric situation the industry needs to work to ensure the greatest commonality in order to maximise the possibilities for intercommunication. The mobile industry needs to look also at the deployment of VoIP services in fixed, private and IT networks, because by being compatible with VoIP, the growth potential of IMS will be enhanced.

In the on-line directory example described in Section 3.2, the user with an IMS-capable terminal in an IMS network will be able to use his rich integrated on-line directory to call his contact. This will be the case even if his contact does not have an IMS terminal or his contact's network provider has not upgraded to IMS. The IMS phone, integrated directory application and SIP call control can set up a legacy call through the IMS network by invoking an E.164 number. However the rich, person-to-person multimedia call[7] would not be available to either subscriber if only one of them is in an IMS environment. Thus the user experience will be greatly diminished unless there is a concerted roll-out of IMS.

8.4 Standardisation Aspects

The progress of the standardisation programme was reviewed at 3GPP meeting #15.[8] Standardisation aspects will be sufficiently comprehensive in Release 5, which will be complete in June 2002, to enable the IMS core network to be developed and deployed. However, there are many other features which are needed for a complete end-to-end IMS system and these are expected to be provided in the next Release which will follow about one year to eighteen months later. These additional features include important aspects such as security, telecom management, charging, header and signalling compression.

A significant number of these features also require action by the IETF. 3GPP has had to apply determined advocacy to ensure that IETF allocates sufficient priority to the 3GPP issues. However, from the status reports, it seems that these aspects will be delivered at the appropriate time and there are no major areas for concern.[9]

The key standardisation issues are summarised in Section 9. These should be discussed with 3GPP, but also in a wider industry context, particularly with the operators who would have to provide the investment for the deployment of IMS. It was also decided at 3GPP meeting #15 that a workshop should be held to discuss the future evolution of UMTS and future releases around the period of meeting #17, with the agenda discussed at meeting #16. Many delegates advocated that a more "top-down" approach should be employed for the future programme. The need for this may be highlighted when the issues surrounding IMS are presented and discussed in open debate.

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[6] Linear growth occurs when a user can intercommunicate with an already large installed base of compatible terminals. An example of linear growth is early cellular phones that could communicate with the massive installed base of fixed phones right from the start. Mobile videophones will be an example of geometric growth, because there will be no installed base of compatible terminals to start with.

[7] In the example this enabled the sharing of a directory, setting up a group call together, breaking off for private chat, multimedia sent simultaneously to both terminals and an e-commerce transaction, all with integrated intuitive, seamless, control.

[8] 11th to 14th March 2002 – Korea

[9] Detailed information on the status can be found on the programme management pages at http://www.3gpp.org. The corresponding information for 3GPP2 can be found at http://www.3gpp2.org.

9 Conclusions

This Annex has identified the following technology requirements for IMS. Not all are strictly "technology enablers", but all require significant technical resources and investment:

- IP-based core network incorporating six new platforms and with backbone packet transmission meeting VoIP QoS requirements.
- VoIP-capable inter-PLMN Internets.
- Concerted roll-out of networks, including IMS, fixed and private VoIP networks.
- Concerted introduction of SIP call control in terminals and networks, with SIP extensions for mobility and security.
- Innovative services which take advantage of IMS, especially interaction and integration of services including presence, person-to-person, person-to-machine, real-time and non-real-time features.
- Toolkits for service creation by third party entrepreneurs incorporating tools for mobility, security, privacy, authentication, registration, billing, service management and integration.
- Telecom Management for IMS platforms and SIP call control, including terminal management capabilities and charging.
- Highly featured terminals including compact browsers, voice browsers, voice recognition, multi-window video displays, innovative audio, pointing devices, memory, storage and downloadable client software.
- The IMS SIM (ISIM).
- The Wideband AMR speech codec is mandatory.
- Significant network tuning and optimisation including special test equipment.

In addition, the mobile industry needs to decide upon a deployment policy, either collectively or as individual operators, for some specific technology enablers. These include:

- IP transport in the RAN (backhaul links).
- "IP-based RAN" (efficient IP over the air including techniques such as Robust Header Compression and Signalling Compression).
- RAN capacity-enhancing techniques such as High Speed Downlink Packet Access.
- SIP Proxies in the core network.

The analysis of the essential technology enablers in this study has revealed the following issues that need to be addressed. Some relate directly to the technology. Others are commercial issues that are opened up by the very different nature of the technology and what it has the potential to provide:

- Should IMS core networks be deployed before the other enablers necessary to provide IMS services are available? Should they be deployed instead of circuit switched networks for basic voice services?
- Will Internet links between PLMNs be available which are capable of supporting VoIP and other real-time IMS services?
- Will the transmission in the GPRS core need to be upgraded to support IMS QoS requirements?
- Should the industry revert to a more concerted approach to network roll-out so that person-to-person IMS sessions can be provided for subscribers who are in different PLMNs or fixed networks and private networks?

- Will IPv6 be available and deployed in time to enable the wide use of Internet telephony as well as the many other 3G applications?
- Should the mobile industry allow external service providers to connect their own multimedia call control and applications platforms to the network so that they can provide their own innovative telephony services similar to the Internet model for information and entertainment services?
- To what degree should the introduction of SIP services and their roll-out be co-ordinated by the industry to ensure a common look and feel by person-to-person users and to maximise growth potential?
- Will service development toolkits that incorporate the necessary additional features for mobility be available in time to allow the development of attractive and innovative IMS services?
- Which legacy supplementary services should be replicated by IMS networks?
- Should the deployment of SIP proxies in the networks be considered in order to gain a short-term advantage but at the expense of potential longer-term disadvantages?
- Is the loss of spectrum efficiency resulting from the need for IP transport over the air acceptable to operators? Will it be possible to upgrade existing 3G RANs to incorporate such technology (e.g. ROHC, HSDPA)? Will the currently proposed techniques provide sufficiently improved spectrum efficiency or is more research needed?
- Should support of efficient IP over the air be mandated in terminals from the start, so that spectrum efficiency improvements can be fully realised when (and if) the infrastructure is upgraded?
- Is the access network independence that can be provided by IMS the best model to suit operators' business plans?
- Will sufficiently powerful browsers be available for small handheld terminals to enable users to enjoy the full advantage that IMS brings to services?
- Should the industry specify certain minimum terminal capabilities for IMS to ensure full enjoyment of the services by users?

IMS potentially offers very compelling new service possibilities. However there are many issues to be resolved which result from the nature of the technology. It is now clear that there is significantly more to implementing the full capability for IMS services than was apparent when IMS was introduced as a core network concept in 3GPP. Almost every aspect of the 3G system is affected and whereas 3G was intended to be an evolution of GSM and GPRS, IMS is more revolutionary. The mobile industry will have to decide how the need for essential new technology fits in with business plans and the current investment scenario, paying due regard to the deployment of similar services in other parts of the telecommunications and IT industries.

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11 References

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3. Extensive reference has been made to papers presented at the 3GPP Services Deployment Workshop and the 3GPP Future Evolution Workshop as well as to 3GPP TS 22.228 and TR 22.941.