Network Convergence or Divergence?

A service perspective on the underlying requirements of future handsets.

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Abstract

Network Convergence or Divergence, what is it, and which are the most likely developments? Today, a much talked about area of telecommunications is the move toward the IP Multimedia Subsystem (IMS), and with it the introduction of IP-based communication in both Public Land Mobile Networks and Public Switched Telephony Networks. This thesis attempts to show whether these two networks will converge with each other and with the Internet in the future - based upon the existence of a few important prerequisites in terms of standards and protocols, whilst divergence in the networks could result if technical and economical obstacles are difficult to circumvent or to exclude from future networks.

The major drivers for convergence include the existence of standards for packet-based transmission over a wide variety of underlying networks, the existence of services that make use of these underlying networking protocols, and ultimately also the existence of devices that can use the new features of a converged network architecture.

The focus is on the impact convergence services or offerings may have on handsets; specifically which protocols need to be supported, as well as those hardware and software requirements that need to be catered for to enable convergence in the handset sector. The thesis concludes with a summary of the most important factors for convergence in future mobile handsets.

Sammanfattning

Nätkonvergens eller divergens, vad är det egentligen, och var är utvecklingen på väg? Idag talas det mycket om inom telekommunikationsbranchen om steget till IMS (IP Multimedia Subsystem), och i med det övergången till uteslutande IP-baserad kommunikation i både Public Land Mobile Networks och Public Switched Telephony Networks. Jag ämnar i den här framställningen visa att konvergens mellan dessa två nät samt ett tredje; Internet är avhängigt av ett par viktiga grundförutsättningar, som existensen av protokoll och standarder som främjar konvergens, medan en divergerande utveckling är resultatet av tekniska och ekonomiska förutsättningar som är svåra att kringgå i framtida nät.

De stora drivkrafterna för konvergens är existensen av standarder och protokoll för paketbaserad överföring av data över skiftande nätarkitekturer, existensen av tjänster som stödjer dessa protokoll, och slutligen användarterminaler som förmår utnyttja dessa tjänster utvecklade för en konvergerande nättopologi.

Fokus för min rapport är på det stöd som behövs i terminalerna för att en sådan här utveckling ska kunna äga rum. Kapaciteten i form av protokoll som dessa terminaler måste stödja, samt vilka hård- och mjukvarukrav som måste uppfyllas kommer också att behandlas. Slutligen kommer jag att framställa en sammanfattning av vad jag anser vara de viktigaste faktorerna i framtida terminaler för att driva den här utvecklingen.

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1. Introduction

Mobile devices have become more and more sophisticated; the trend today is to have phones equipped with many different types of interfaces, for example WiFi, GSM/GPRS/EDGE, and 3G. Simultaneously, there are extensive ongoing standardization efforts concerning mobility and interoperability based upon standards such as SIP (see section 1.3.1.), Mobile IP (see section 1.3.2.), H.323 (see section 1.3.3.), UMA (see section 1.3.4.), etc. These standards attempt to provide mechanisms for maintaining existing media sessions while supporting the handset's roaming between different networks. Together, these prerequisites have resulted in a platform for new types of services that build on the increasing convergence of telecommunications and data communications. One such service made possible- by SIP alone, is the "one phone" concept: having one address identifier although a user might have multiple devices and points of network attachment. Another service facilitated by SIP is presence indication, i.e., whether you are available for a chat or busy.

As the different networks converge, due to adoption of internetworking protocols which seek to harness the rapid innovation of the Internet community, we might soon see some very interesting new services. If, however, convergence is not viewed as advantageous by all parties involved in this transition, or not implemented in all networks with a view toward interoperability, this will hamper convergence and delay those benefits that can be brought about through use of such convergence. This probably would mean there will be fewer new services, some services only working in some networks, incompatible implementations of services etc.

The drivers for convergence are also inherent in the development of the underlying network architecture, for example the introduction of packet switched nodes in mobile networks (e.g. GPRS Support Nodes), which utilize packet-based core networks. Should this development continue we will probably see packet-based networks, often called All-IP networks, where the routing of IP datagrams is the basis for communications, while the legacy circuit-switched parts of the existing networks will ultimately be phased out.

Moreover, what we can clearly see in the marketplace is a trend toward convergence between three big industries: media, telecommunications, and IT [22]. The media companies want partners which can provide them with new and secure channels to reach their customers. Thus, the telecommunications companies seek to provide them with the possibility of reaching a vast number of end-users with content through a secure channel. These new ways of providing content is an attempt to counter the losses that arise from pirating of traditional media, (for example mass copying of CDs) meanwhile they represent a market opportunity to reach a greater number of potential customers [33]. That is one reason why new services such as streaming music and video are being introduced to mobile telephony customers, in addition to legacy services like voice and text messaging.

These new services can possibly also have a large effect on mobile operators' revenues, as the price of voice services has continued to plummet due to fierce competition from other operators, regulatory statutes, and now also the threat of new competition, primarily due to Internet Telephony, also called Voice over IP (VoIP). Consequently, many mobile operators are looking both to safeguard the revenues they already are generating through circuit switched (CS) voice and at the same time to find new sources of revenue.

Already today fixed VoIP extensions uptake exceeds the increase in number of business PBX extensions [43]. A clear concern for mobile voice operators is what effect mobile VoIP and use of local wireless connections will have in diverting voice traffic away from their mobile networks.
IT managers are closely involved in this development, wanting to see solutions that preferably make use of the installed base of computer equipment in large and small corporations, where services like e-mail and remote corporate access, or even real-time inventory of goods are routinely provided smoothly to roaming employees; their desire is to do the same for voice, videoconferencing, etc.

Subsequently, more and more services are being launched, either by the mobile operators or by independent service creators which build on the concept that the functionality of a mobile device can be extended beyond voice and text messaging. Of great interest is the question of which of these new services will make it “big” (for example, producing revenues similar to what SMS produces). It will ultimately depend on a number of factors, but ease of use, operator launch support, and adherence to global standards appear to be some of the most important aspects in marketing a new service [22]. New mobile applications need to be intuitive and interoperable with one another to stand the test of predominantly young customers, scrutinizing every new thing with harsh judgement; otherwise they might quickly dismiss the service.

However, there seems to be no single technology or standard which will be the basis for every new convergence offering we will see in the mobile devices of tomorrow, because different services have differing requirements in terms of capabilities and functionality. While this need not necessarily hinder development, interoperability between independent versions of similar services would surely benefit from a more harmonized underlying structure. However, some services may be able to “make it on their own” independent of the underlying network structure, simply by drawing attention to them due to a globally recognized brand associated with them. One such example is the recently launched “Google Maps for Mobile” application [66], and this factor is also one that should not be underestimated.

Altogether, these characteristics make the telecommunications sector an interesting and complex market where services and the capabilities of the devices, are spearheads driving convergence of the networks: fixed, mobile, and the Internet. Some important prerequisites for convergence services to be attractive in the eyes of the consumer would most probably be that they can make use of different networks to provide customers with either the cheapest available bit rate, or the best-suited connectivity for specific applications at a particular place or time.

1.1.1 Objective of this thesis

To provide some background for examining the issues of convergence, the subsequent part of this thesis will discuss some of the relevant protocols used for supporting mobility in the Micro- and Macro-domains, specifically in the latter case between areas of different (overlapping) networks, such as WLANs and 3G/GSM, and in the former between homogenous network segments. The third part of the thesis discusses the inherent characteristics of the wireless access networks that provide the basis for the new services will be treated. In the last two sections, the current and future convergence service offerings originating either from mobile telephony operators or from other service purveyors will be discussed. The trend of Internet applications "going mobile" as well as a discussion of the general impact of the new services on component developments which play a big part in determining the future hardware capabilities in the handsets, and the services they can subsequently be expected to support are likewise to be found in the subsequent parts of this thesis.

The final section of this thesis seeks to determine which could be the most important drivers for promoting convergence of the different networks, and what could be the major obstacles hindering such a development. One part of the final result will be a synthesis of these developments and how they translate into capabilities required in future mobile terminals. The question that the thesis ultimately seeks to answer is if we are moving toward convergence or divergence in the mobile and wireless landscape and what this means in terms of capabilities needed in future terminals.

1.2. Telecommunications sector standards development

In the telecommunications sector, new technologies and upgrades to existing networks have traditionally been realized through a planned, “bottom-up” structured approach. For example, the way new frequency modulation schemes and frequency bands for usage have been introduced has been through an extensive standards drafting process. Subsequent infrastructure vendor implementations then adhered to these standards, (such as those developed by bodies such as the ITU-T [17], GSM World Association [18], ETSI [19] or, with the advent of GSM & UMTS standardization, the 3GPP [20]). This has provided certain advantages for network operators, in the forms of aligned equipment capabilities, which meant that independent parts of networks architecture could be bought from different vendors and still be expected to offer specific basic functionality and interoperability. Furthermore, this had the implication that mobile network operators could only choose from a predefined range of technology enablers, which combined with poor end-user marketing sometimes resulted in poor uptake of services. Take the example of WAP, which was originally conceived as the “Internet going mobile”, and marketed as such to the customers, leading to a huge failure due to its inevitable, unfavourable comparison with its older cousin, the “real” Internet [22].

Another important force in the development of convergence standards in the intersection of the Internet and the mobile world are standards bodies such as the IETF and the IEEE, who seemingly have adhered to a more “organic” approach than their telecom counterparts. The IETF specifically promotes various interoperable Internet standards in the form of Requests For Comments (RFCs) based upon interworking implementations, which is very interesting from the convergence perspective. These two bodies have evolved from the sphere of computer communications and with the introduction of various wireless data networks the trend is for data communications to move into the telecommunications market. Already we see that data communications networks are able to support real-time applications such as voice or multimedia with a quality equal or better than that of the traditional circuit-switched world [44]. Provisioning for these services often does not involve classic reservation schemes parallel to those of the legacy CS world (which implies dedicated QoS and predefined path setup), but simply providing high enough capacity and throughput for the packets to be routed swiftly between network end points.

Notably a lot of work is going on in the area of data- and telecommunications convergence and this is easily seen by looking at a list of IETF developers, where it is common to see people from competing equipment providers, such as Cisco, Ericsson, and Nokia working together along with a wide spectrum of other participants on convergence focused standards. A good indication that we are moving in the direction of a convergence of networks is that we see many new standards focused on enabling mechanisms for mobility between different (wireless or wireline) networks.

1.3. Mobility protocols

In order to understand which factors facilitate mobile service offerings, one needs to look at some of the communications protocols that make it possible to roam seamlessly from one network to another, while maintaining a stable session with a correspondent node (CN). These communications protocols, be it on the network level such as Mobile IP, or the session level such as SIP, are referred to in this thesis as mobility enablers. The thesis will discuss both macro mobility, which makes it possible to perform so called vertical handover **between** networks of different types, and micro-mobility, which makes it possible to roam **within** a particular communications network (in second generation mobile networks for example this is accomplished through the use of GPRS Tunnelling Protocol (GTP) for General Packet Radio Service (GPRS)). The modified OSI model shown in Figure 1 shows the location of some communication protocols providing mobility (they are shown in blue letters on a white background), and I will discuss each of them in turn in the following sections. On the third highest-level, we have the session management protocols, which ensure that communications sessions between peers can be set up and maintained for the remainder of the session. Here we find SIP, which has evolved to become an important enabler protocol for the provisioning of new, integrated services between the IP and the telephony worlds. This is further discussed in section 1.3.1. At the IP layer, the most important protocol for providing mobility has so far been Mobile IP and to some extent also H.323. Mobile IP and H.323 are discussed in sections 1.3.2 and 1.3.3 respectively. A short introduction to the work of the 3GPP UMA (Unlicensed Mobile Access) working group will be given in section 1.3.4.

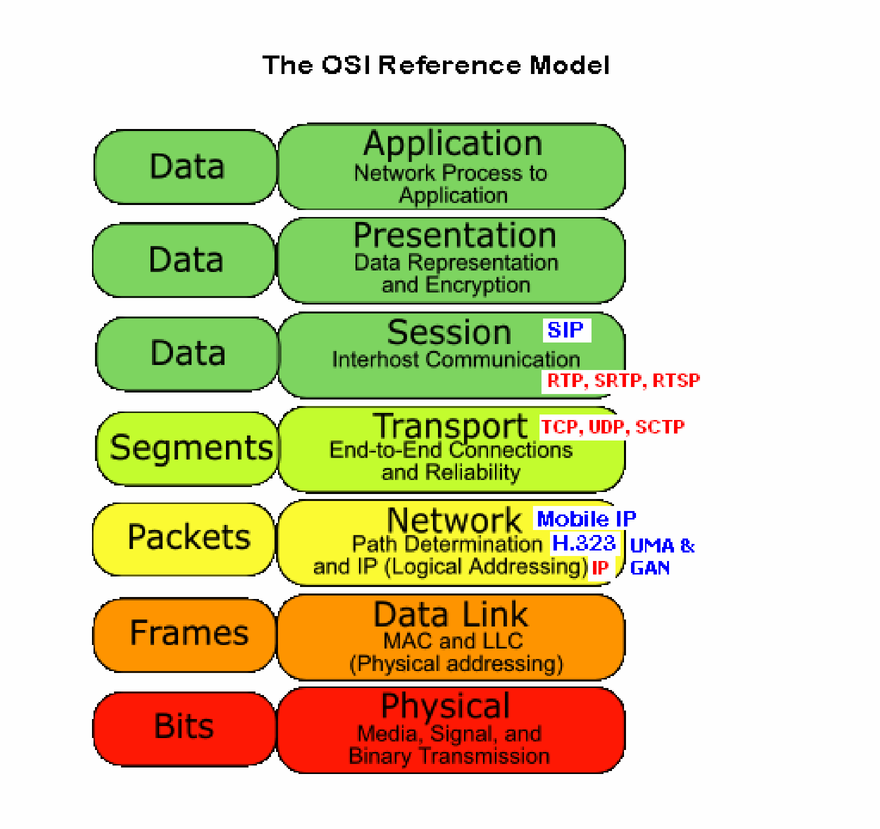

Figure 1 – OSI Model with respect to mobility enablers

1.3.1. Session Initiation Protocol

The Session Initiation Protocol is a text-based application protocol that allows endpoints on the Internet to discover one another in order to exchange information about a session they would like to create, modify, or terminate. SIP entities describe this session using the Session Description Protocol (SDP). SIP can support Internet telephony, Instant Messaging, video, Internet gaming, and other forms of real-time and non-real time communications. The adherence to and the popularity of the SIP protocol is in many aspects due to the fact that its structure is very similar to the HTTP protocol, which due to its *simplicity* gained such a large following of developers in the Internet community. Today SIP is gaining momentum as the enabler for future IP-based services in 3GPP’s IP Multimedia Subsystem (IMS), a standard which many large operators will follow. This includes traditional telephony operators such as Telia and Tele2, who already provide IP-telephony built upon the SIP architecture [2][3]. Having seen the effect of VoIP on many traditional fixed telephony operators, the mobile operators are following what is happening in this line of development carefully. Not surprisingly SIP has been chosen as the common framework for introducing new multimedia services in IMS, and this means that most probably it will be an important enabler and a driving force for the mobile Internet in the years to come.

1.3.1.1. SIP addressing scheme

A SIP address has the form of a Uniform Resource Identifier (URI) of the form username@somedomain. The username part of the address could be a mnemonic name, (such as Jacob.Possne) or it could be a string like ZXTRQS157, or it could have the form of an E.164 number: tel:+4687906000, which subsequently can be translated into a valid IP-address through the use of ENUM [10] procedures. The domain name part of the identifier is linked via a DNS record specifying the location of the SIP Proxy specific to the domain part of this user’s SIP provider.

A user can have several terminals associated with a single URI, which makes it possible to use a concept called *forking* which is explained below. This means that for an incoming (media) session request, the user can choose whether the call should be *sequentially forked* or *parallel forked* to his or her devices. In the first solution each of the terminals in turn will be contacted by the user’s SIP Proxy, and queried in sequence for acceptance of the media session. In parallel forking, as the name implies, all terminals will be queried at the same time, allowing the user flexibility to select the type of communication he wants to set up based on the device’s capabilities. Figure 2 on the next page illustrates this situation.

1.3.1.2. SIP call set-up

When setting up a call, or more formally a media session, SIP employs the SDP protocol for parameter negotiation. The procedure is depicted in Figure 3, and it is outlined below.

The user that wants to set up the call sends a SIP INVITE message to their outbound proxy (in this case the atlanta.com SIP proxy) The message contains some header information which includes a unique identifier for the call, the destination address, Alice’s address, and information about the type of session that Alice wishes to establish with Bob.

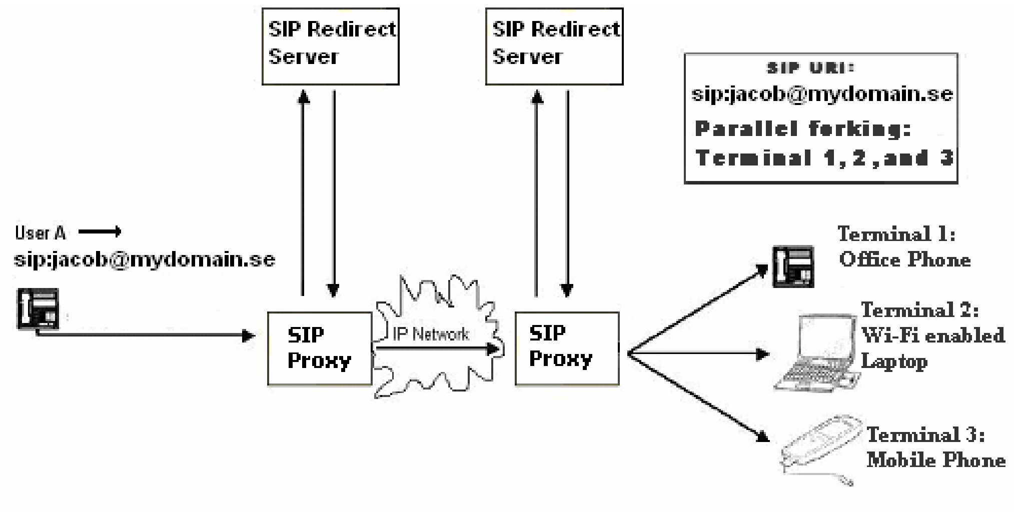

Figure 2 – SIP forking

The proxy server, which might have been discovered by DHCP or preconfigured in Alice's softphone replies with a "100 Trying" message, which indicates that the message has been received, and that the proxy is working on the caller's behalf to route the INVITE to its destination.

To route the INVITE to the SIP server of the domain biloxi.com, Alice's proxy server possibly has to perform a DNS lookup, normally a DNS SERV [11] which translates the SIP URI into a collection of IP-address, port, and the supported transport protocol of the inbound Proxy server at biloxi.com.

When the SIP Proxy at biloxi.com receives the INVITE it replies with a "100 Trying" message to Alice's proxy, and looks up Bob's currently registered location via a location service (explained below) to pass on the INVITE to the current IP address of one (or more) of Bob's SIP User Agents (typically co-located with each SIP device).

Finally Bob's SIP phone receives the INVITE and starts ringing, and at the same time sends back a "180 Ringing" message to its SIP Proxy, which removes itself from the Via header field, and passes the message on to the next entity in the reverse path in the same header field.

When the "180 Ringing" message reaches Alice's softphone, she knows that the call has been appropriately routed to its destination and subsequently waits for Bob to answer.

When Bob picks up the receiver his SIP User Agent sends back a new message "200 OK" with a message body that contains the media description that he is willing to establish with Alice.

Alice's softphone answers with a "ACK" and at that moment the media flow begins, note that it will takes a route directly between the endpoints, as the endpoint addresses were included in the SDP message bodies of the INVITE/OK messages. The important thing to note here is that the media and the signalling need not take the same path, which is often referred to as the SIP trapezoid (See Figure 3).

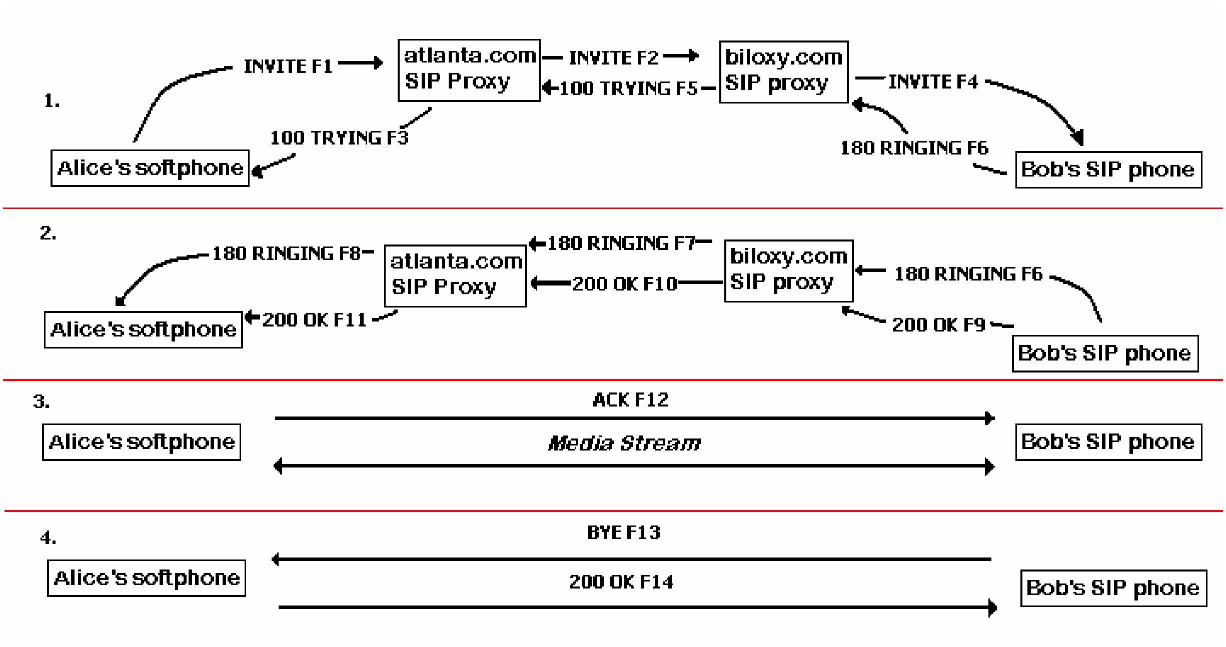

Figure 3 – SIP Trapezoid

1.3.1.3. SIP Network elements

SIP works in conjuncture with a number of servers, each of whose functionality is described in the following paragraphs. Note that the distinction between servers in SIP is purely logical, thus, they could coexist in a single physical entity.

1.3.1.3.1. SIP Proxy Server

A SIP proxy server is the gateway that forwards requests on behalf of SIP clients, when they attach to a SIP domain. If a secure SIP URI of the form sips:xxxx was used [1], then the proxy ensures that an appropriate security mechanism is used between the caller and the called party for transporting the SIP signalling. The transport protocol used is Transport Level Security (TLS). The proxy forwards or rewrites the original requests from the client to the next server, such as a redirect server or a registration server. If a secure SIP URI was used, then each of these pairs of SIP entities should also set up a TLS session between them before forwarding the SIP signalling.

1.3.1.3.2. SIP Redirect Server

A redirect server handles redirection of requests it receives to direct the request to an alternate URI, which then handles the request. The redirection server takes itself out of the call path after aiding in locating the target of the request.

1.3.1.3.3. SIP Registrar Server

This SIP Server receives registration messages when a user attaches to a network and subsequently when receiving a renewed registration request after a predefined interval. The registrar stores this registration information at a *Location server*. The location information is called a binding and the registrar server's purpose is similar to those of the HLR and VLR databases in mobile networks.

1.3.1.4. SIP as an enabler for new convergence services

SIP is a session layer protocol and as such it can use any transport protocol that is supported by all the end nodes in a communications session. The session may utilize common protocols such as RTP, SRTP (Secure RTP), or RTSP. These protocols are commonly used for streaming content as music or video to devices. RTP for example can be used to carry encoded audio, video, or even Timed Text (used for providing timed text to a video sequence as defined by the 3GPP, see RFC 4396 [4]). Media can also be added to or removed from a session in progress through SIP, making it agile to changing communications environments. A SIP re-invite during an existing communication session enables the session to continue even while a session with new parameters is being set up.

The inclusion of presence indication in SIP is something which could make it attractive for new forms of Instant Messaging solutions (at the moment, emerging in the market is the well-known MSN Messenger for mobile devices [26]). It works quite simply by the caller, which we call A, sending a subscription request to the proxy of the called party, B, to see B's status. At B's proxy, however local policies can be applied for an already registered user to accept or deny status-requests by correspondent nodes, similar to what ICQ users have come to know as "appear offline".

Another defining feature of SIP is that it works independent of the underlying transport-level protocol, so whether the user is using UDP, TCP, SCTP, or any other transport protocol is unimportant from the SIP architectural point of view. Additionally, as SIP assumes no responsibility for the reliability of the media session, but only takes care of the signalling for the media session set-up and tear-down, SIP doesn't care what protocols are used for the media, it simply passes on the initiating user agent's list of protocols to the other user agents. Thus, protocols such as RTP, RTSP, or SRTP can be used to carry the media session. Because of the fact that SIP is neutral to the transport protocol used, it is also neutral to the network layer protocol, thus Mobile IP or another network protocol can be used to support mobility.

1.3.2. Mobile IP

Mobile IP [41] provides network mobility for end-nodes at the IP-layer. Thus the roaming subscriber who attaches to a new network, and possibly also receives a new IP-address, can still be reached using his home address. This is made possible through the use of so called Home Agents (HA's) and Foreign Agents (FA's); when a subscriber moves to a new location, the node attempts to register with the advertised FA present on the link, if there is one, and then it sends a registration message back to his home agent. If there isn't a FA on the link, then the Mobile Node (MN) must act as a FA itself using a so called co-located care-of address, which is a network address from the new network, associated with one of its interfaces (the interface that is connected to the new network). If packets are sent to the MN by a correspondent node unaware of the MN's mobility, the packets are routed to the nodes home address, where they will be intercepted by the HA and forwarded in a tunnel to the Care-of-Address with which the MN has registered. The MN can then reply either directly or via a reverse tunnel.

The first method of replying to the CN will only work provided the CN doesn't reside in a network which performs *ingress filtering* (i.e. checks whether the source address is topologically from the right network to avoid DoS attacks which employ address spoofing in order to impersonate other network entities, see Figure 4)[6].

The second method will always work provided that the MN sets up a tunnel to its HA, that the HA has a publicly reachable IP address, and that the MN doesn't reside behind a network address port translation (NAPT) gateway which causes loss of reverse path information. The way this latter problem occurs is that when the NAPT-gateway (which is commonly also the FA) transmits a message from the MN, that communication stream is multiplexed and mapped to a single external identifier. Some important information such as the port number for the communications stream is subsequently lost, which makes the reverse path determination impossible for the HA, as it can only see the public address of the NAPT-gateway, and not the port number, so in that case reverse tunnelling fails. Irrespective of if reverse tunnelling is used or not, this is problematic from the MN's point of view, as the media session will be hindered by the loss of information to the HA occurring at the NAPT, so this could potentially lead to trouble in maintaining a communication at the network layer (See figure 5).

A mechanism that can be used to remedy this situation, however, is the use of UDP in IP tunnelling, as described in [5]. This mechanism works by setting one bit in the UDP registration message to the home agent; indicating to the HA an address through which messages can always be routed back to the MN, i.e. IP-address and port number. To this address the HA forwards subsequent IP-datagrams, delivered through the NAPT using the very same destination port as was used for registration by the MN, for the remainder of the communications session.

From this discussion we can see how Mobile IP is well adapted for use in maintaining sessions independent of the underlying access network. However, the delay in registration with the HA and potentially also the FA when performing a handover from one network to another, or changes in throughput at the new network, will determine the success of maintaining an existing communication session. These dependencies will be more thoroughly investigated in the section about effects of vertical handoffs (see section 2.3.) on VoIP performance, where in all cases Mobile IP was used as the network mobility protocol.

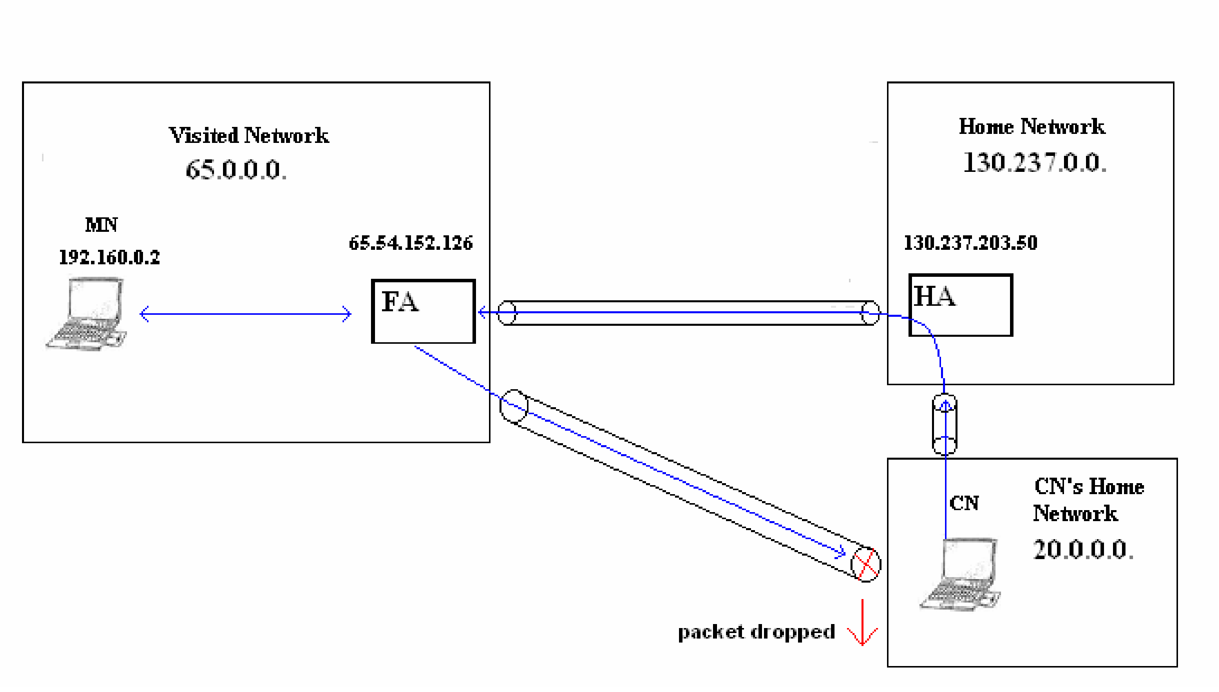

Figure 4 – Ingress Filtering

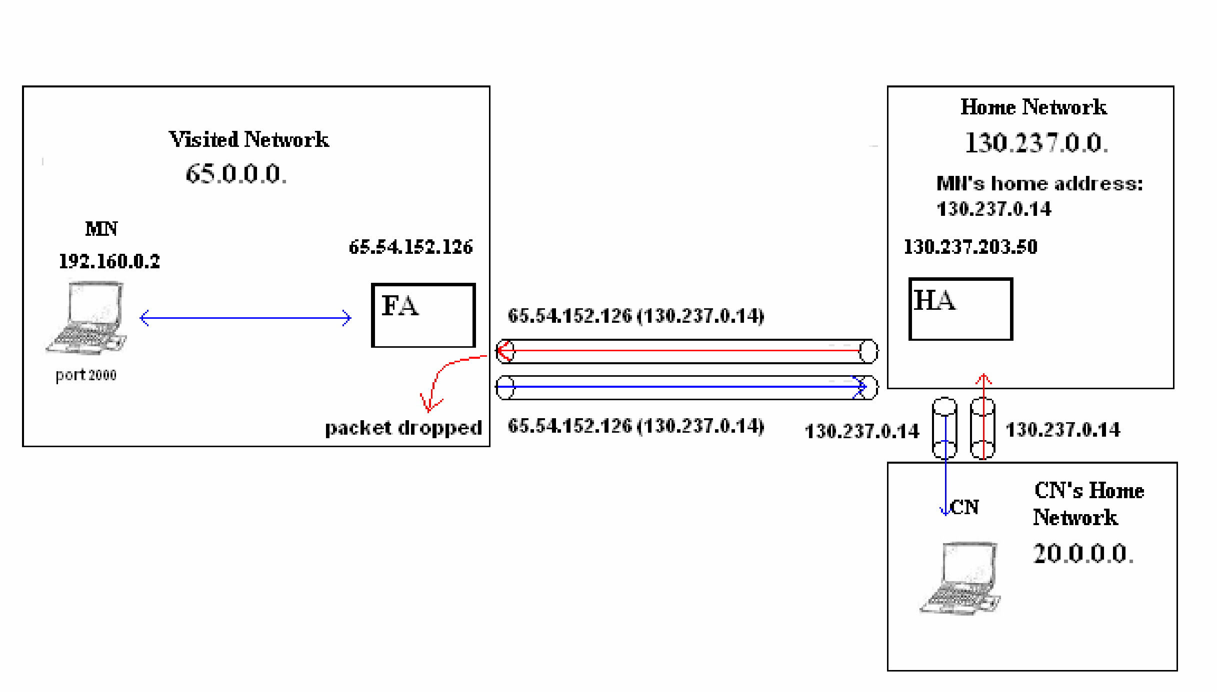

Figure 5 – Failed reverse path tunneling

1.2.3. H.323 and the TIPHON working group of the ETSI

In 1997, the ETSI started to work on Telecommunications and Internet Protocol Harmonization over Network (TIPHON) to propose specific mechanisms for interworking between different types of networks. A short term goal was to promote the H.323 protocol for providing VoIP via packet-based networks (e.g. IP networks) and interworking functions with mobile networks. H.323. was originally proposed by the ITU-T [8] in 1996 (with no QoS guarantee) as a protocol to provide IP telephony that supported the signalling of traditional telephony systems. The last amendment to the protocol was made as late as in 2005. An H.323 network has various nodes that are similar to the nodes in a mobile network, and perform similar tasks such as location updating, call signalling and setup; and employs the use of protocols such as GSM MAP, SS7, and ISUP/TCAP in negotiating call setup with legacy PSTN and GSM networks.

H.323. will not be discussed further in this report due to its status being as primarily a legacy protocol; currently the TIPHON body has been merged into the TISPAN (Telecoms & Internet converged Services & Protocols for Advanced Networks). Additionally Microsoft, who for a long time delivered a copy of their communications solution “NetMeeting” with each copy of their Windows operating system, has switched from using H.323 as a basis for this application to using SIP instead [9].

1.2.4. UMA and GAN

Unlicensed Mobile Access (UMA) is a standardization effort, which started in 2004, and in 2006 merged with the 3GPP and is now called the Generic Access Network (GAN). A first “user perspective” document was written in 2004 where a new and converged networks architecture was outlined. However, the problem of authenticating users and terminals in UMA had to be addressed, as WEP (Wireless Equivalent Privacy) was used in this technical report for authenticating end users, and the fact that it had been broken by Stubblefield et al. [24] wasn’t taken into account. Thus another end-to-end security protocol such as SIPS for authentication could be used in order to achieve adequate end-user security. However, it should be kept in mind that every added step in authentication adds to the overall delay, making the VoIP handover longer, and potentially leading to degraded call quality if the delays are too long. When UMA merged with 3GPP, it adopted the previously developed GAN approach for allowing seamless handover between different wireless networks; thus when the handset is able to communicate via a WLAN or a Bluetooth access point, it establishes a secure IP connection through a server called a GANC. It thus appears to the core mobile network as if the subscriber is simply using another base station. This has several implications; one is that battery life may be reduced when the device is equipped with several interfaces that are powered on simultaneously (but only some are active), another is that data still has to pass through the mobile network for registration and location awareness. Hence, this data will be charged for, even if the mobile service provider and the broadband service provider possibly are the same entity, leading to additional cost for the end-user. A benefit of this new scheme, as of [39] is however, that secure tunnelling is used to connect to the GANC, using any key generation method of EAP-SIM [42] or EAP-AKA [42], employed with authentication through IKEv2 [39]. Then session encryption between the communicating entities is similar to how it is done in mobile networks, through the use of a session key, K_c . 3GPP, which is an ETSI body, should be expected to have a large industry following, so this approach should not be disregarded.

2. Access Network Technologies

Wireless networks will be significantly affected by a convergent architecture, where the end node can switch seamlessly between different forms of connectivity. Therefore a short introduction to the basics of communication over these networks in terms of their inherent structure and their behaviour in terms of bandwidth, medium access, and other characteristics is given.

2.1. Wireless LAN

The IEEE 802.11 standard for wireless LAN was proposed by the IEEE in 1997, as a Direct Sequence Spread Spectrum technique for utilizing radio resources. The standard was designed for operation mainly in the unlicensed Industrial, Scientific and Medical band (today, mainly the 2.4 GHz band is used). It has seen a huge uptake at sites such as campuses and for private residential networks, due to its comparatively low acquisition cost. Consortia or communities have formed which promote the sharing of wireless access especially in urban areas (see www.seattlewireless.net or www.elektrosmog.nu). Originally the bandwidth supported was only 2Mbps, later the 802.11b standard was amended to handle data rates of up to 11Mbps, and later, with the introduction of 802.11a and 802.11g standards; support for up to 54 Mbps was added. In practice, however, the throughput for the most commonly used 802.11b standard is lower than the maximum user rate; according to Syed [16] the typical throughput is 2Mbps, due to the fact that for the higher data rates a lot of physical layer overhead added [16], plus the fact that transmission error rates can be as high as 50% at times [16]. IEEE 802.11b defines two ways to access the medium, one is a Distributed Control Function (DCF) and the other is a Point Coordination Function (only usable in infrastructure mode). The DCF function, which is most commonly used for infrastructure mode, defines a media access control protocol (MAC) which employs Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). CSMA/CA is an algorithm with a random backoff timer to avoid collisions in the radio channel. Each node in the network has a backoff timer, which is initially set to a random value between 0 and the minimum contention window size, CW_{min}. When the timer, which decrements every 20μs, reaches 0 a frame can be sent out on the radio channel if the channel remains free after the DIFS interframe space has passed, otherwise the retry counter is incremented by one, and the contention window (backoff timer) is doubled. This process will continue until the contention window reaches maximum size, the retry counter reaches its maximum, or the time-to-live of the packet has reached 0. The medium access protocol in WLAN is thus an important characteristic which determines the delay before a node can transmit, and can influence throughput in the network. Handover between adjacent WLAN cells connected to the same backbone subnet is typically determined by the characteristics of the WLAN interface. Some manufacturers use a low threshold for switching between APs in terms of signal to noise ratio measurements at the physical layer. Syed, in his thesis investigated the effects of handover between WLAN access points and defined a classification for WLAN interfaces [16]. He divided them into *Network/Hotspot* friendly or *Single AP/Home User Friendly* based upon their respective handover behaviour. Thus, interfaces which followed the first behaviour would rapidly switch between access points, while clients with the latter behaviour tried to use an access point for as long a time as possible (basically until they completely lost connectivity via this access point. He also showed that for many VoIP clients, the handover delay was directly responsible for the VoIP client's ability to maintain existing communications sessions. These results will be commented upon further in section 2.3.

2.2. GPRS

With the surging popularity of the Internet and its packet-switched services, ETSI and other standardization bodies saw the need for the introduction of such services into wide area mobile networks. Thus in 1994 the first standardization efforts in this area began. When GPRS was conceived, it was designed to reuse the existing GSM infrastructure, so only minimal upgrades would be necessary by the network operators. However, as radio resources are a limiting factor in GSM Networks, GPRS capacity is determined by the number of timeslots that can be assigned per channel, and the data rate for a given encoding and error correction level per time slot. The introduction of E-GPRS increased the spectral efficiency allowing additional encodings and error correction levels in each time slot, which in practice means ~144 Kbps outdoors and ~470 Kbps indoors[13]. Third generation WCDMA technologies however, promise data rates of at least 384 Kbps when a user is moving about outdoors [13], and WLANs typically provide at least ~1Mbps. Therefore GPRS should arguably be viewed as a transition mechanism for mobile operators who wish to introduce packet-switched services without requiring a substantial investment or change in their network. Additionally, the GPRS core network and IP backbone architecture can be reused, when deploying for example WCDMA access nodes. GPRS can thus be seen as an important step in a transition to IP-based services and per packet or per data volume charging for traffic. GPRS was also designed to coexist favourably with such protocols as TCP and/or UDP; actually the error correction mechanisms of GPRS and TCP have been found to coexist very well [14]. However, the initial GPRS encodings and error correction mechanisms were not suitable for delay sensitive packet-switched services, such as voice. The reason for this is that GPRS was originally designed as a complement to the existing circuit-switched voice services. Thus, GPRS adds error-correcting mechanisms that increase overall delay (see Figure 6). In reality, GPRS is not a pure packet-based protocol, but rather it utilizes network assigned resources. The resources are managed by a Packet Data Protocol (PDP) context (see Figure 6), and the one-way delay of packets sent every 500ms interval is typically ~600 ms [21] or more. Moreover, PDP context activation delays make it infeasible to quickly initiate packet communication, even using the fast reservation scheme. GPRS needs at least 500ms before a PDP context can be activated [13] which is probably the best case scenario when performing a vertical handover with two live interfaces. Thus, the extensive delays in registration and limited throughput of GPRS are two characteristic that makes it unsuitable for supporting real-time traffic, where frames are worthless if not received in time; Rosenbluth and Cole determined the threshold before communications quality starts to deteriorate according to the E-model to be ~170ms[15]. Therefore, both the supported delay of VoIP clients and the probability of roaming are determining factors for the number of dropped calls that will occur using VoIP over GPRS.

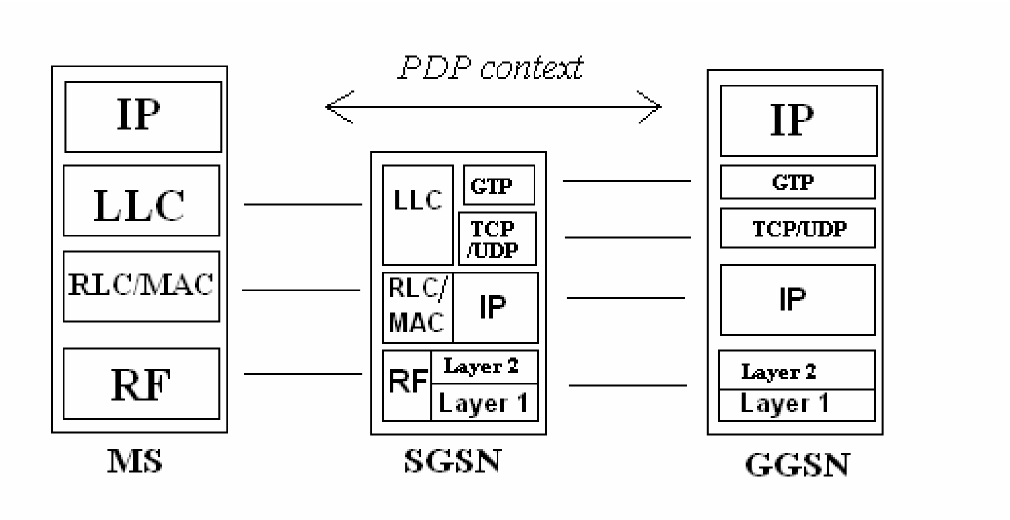

Figure 6 – GPRS PDP Context

Radio Link Control /Medium Access Control Layer

The RLC/MAC layer provides services for information transfer over the GPRS physical layer. This optionally includes error correction based upon selective retransmission of erroneous blocks.

Logical Link Control Layer

Dependent on amongst other things the QoS profile negotiated between SGSN and the mobile node, the LLC layer may perform functions such as error correction and protection of data at the frame level. It is also responsible for setting up a connection oriented or connectionless path between the MN and the GSNs.

2.3. Earlier Work : Effects of handovers on application performance

Behaviour of VoIP clients during horizontal handover (i.e. handover between networks of similar characteristics) was researched by Syed[16] in his thesis. In his tests, the length of the phases when the roaming mobile node starts search for a new WLAN network to attach to after the signal strength has fallen below the threshold, (which Syed calls the *continuous unacknowledgement phase*) and the phase when the MN scans the different frequencies for a suitable WLAN to attach to (*named the scanning phase*) were the phases which affected the roaming clients behaviour the most with respect to terminating or maintaining their communications session. The combined length of these two phases in Syed's investigation was on average ~1400ms; at ~1700ms most of the VoIP clients in his tests terminated the ongoing call. In addition, it has been stated that the delay threshold before an interactive voice session starts to suffer degraded speech quality is around 170ms (Rosenbluth and Cole in [15]), so interactive speech would have been lost long before the session is terminated. Then, in addition to the link layer handoff time we should normally add additional time for authentication and authorization and other registration delays. Thus we can see from this why handovers between WLAN APs requires a minimized delay at network interfaces to be able to support real-time services like VoIP where the delay between successive packets is crucial for both preserving speech quality and avoiding dropped calls.

With regard to vertical handovers, (i.e. handovers between networks of different type) the impact on the throughput of an ongoing TCP session in Wu's study [14] from the application point of view (a file transfer) was that the TCP contention window, which had been set to a comparatively high value during access to WLAN from the start of the roaming, was reduced when the MN performed the so called "upward" vertical handover, from WLAN to GPRS. In addition to this a few frames had to be retransmitted which were sent during the time when the MN attached to the GPRS network, a period that "for lack of no accurate reference" [21] in Chen's thesis [21] was estimated to take about 500ms (consistent with the value in [13] for the fast reservation scheme, which then constituted the entire attachment delay). What is interesting however in this is that in switching from WLAN to GPRS, it took a total of roughly 10 seconds for the file transfer to *adapt* to the changing conditions of the link, while in the opposite case of "downward" vertical handover between GPRS and WLAN the handover adaption was almost negligible. Thus we can see that for delay-sensitive applications such as VoIP this scenario will probably not be so appealing, and that for applications that are focused more on throughput (such as file transfers for eg.) the case of an upward vertical handover is possible[21], but far from ideal.

It should be mentioned however, that the testbed that Wu employs [14] doesn't require authentication for the attachment to the WLAN network, i.e. it used so called "open authentication" where the most important factor affecting the delay for Mobile IP registration messages can be estimated by the time to exchange registration messages between the MN and HA. In GPRS networks, the authentication mechanism is performed on a session basis employing a session key K_c which is exchanged between the assigned SGSN and the MN during PDP context set-up, so consequently; while it can be said that his experimental measurements in some respect is asymmetrical, yet they are very interesting because the increased round-trip time in the GPRS network was determined to be the major cause of the packet-loss during the handover process from GPRS to WLAN, consistent with [40].

Theoretically, however, since voice can be compressed and still retain good quality at 8Kbps[16, 34], packet-based voice **could** possibly be accommodated even at GPRS's lowest sending rate (through the CS-1 RLC coding scheme) of 9,05 Kbps[13] (which incidentally has the most RLC level error-correction). It should be noted that the delay in attaching to the network, (~1200ms [21], if no authentication is used) is still below the 1700ms, when most commercial VoIP clients in Syed's tests [16] terminate the call, so the call could be maintained during the handover from WLAN. Consequently, if one wishes to keep an ongoing VoIP session, independent of the underlying networks through the use of protocols like Mobile IP and SIP, it could probably be done, although the quality that would result from it when roaming within GPRS certainly remains debatable. This possibility was also further researched in my experiments with commercial VoWifi clients section, because of this presumption.

However, we should not fail to take into account that the probability of a vertical upward handover taking place in the first place. Handovers normally aren't necessary if the user is sitting in the waiting lounge of an airport, at his office, or at home. Therefore, it would be interesting to determine the probability that a vertical handover will in fact take place. Horizontal handovers would probably be a little more frequent, since the user might be moving about in an office complex, or in the airport, albeit normally not so much while at home. The probability of horizontal and vertical handover and its impact on VoIP performance is thus an area which could be studied in future research. Later in this thesis, an attempt was to test VoIP performance in two roaming scenarios, running both the Skype™ Mobile client [49] on a HP iPAQ 5550 (see section 4.2.1.1.) to see how roaming between intra-BSS AP's impacts VoIP performance, and the Woize[51] client to test handover and VoIP over GPRS on a Qtek 8310 (see section 4.2.2.2).

3. Convergence services

The existence of the mobility standards in terms of providing mobility and ubiquity of applications independent of the underlying network is of course a very important precondition for future applications in this market, but there are also more general usage trends which shape the market for mobile services. Notably, there is often an interest in taking an already established behaviour or service and to try it out in the mobile channel. Thus a common denominator for many applications, like VoIP, or instant messaging: (for example Agile messenger or MSN Messenger) is that they build upon already established Internet usage patterns and behaviours and apply these to the wide area (mobile) networks. There are many potential services which could be realised as an effect of the convergence capabilities in the devices, and there are yet others, which are introduced in the devices as a consequence of their increasing capabilities, such as broadcast TV reception or GPS navigation.

These different device capabilities and services will be discussed in the following sections; in section 3.1.1. native Internet applications migrating to the mobile community will be elaborated upon. Especially important is VoWiFi, which is available from only a few companies in Sweden so far [46]. Section 3.1.2. will be dedicated to provisioning for purchase, streaming, or playback of music in digital format through wireless communications networks, which is an increasingly important area for both mobile operators and content purveyors. Section 3.1.3. discusses the introduction of commercial TV offerings via both the mobile network and the broadcast network, and its associated technology prerequisites. Finally, the last section of this chapter will delve into the possible implications of ubiquitous gaming service offerings, with special focus on the multiplayer aspect; when a player is switching between different types of access networks and wishes to maintain an ongoing gaming session with a number of other players.

3.1. Native Internet Applications on cellular handsets

In the mobile marketplace today, we see more and more traditional Internet services: web-browsing, e-mail, and Instant Messaging, not to mention, of course the present hype around mobile VoWiFi[47]. Solutions such as the one supplied by Optimobile [47], Skype™ for Mobile [49], or Woize™ [51], make it increasingly appealing for end-users who wish to use their handhelds for VoIP when in either GSM/3G networks or WiFi environments, hence they can make use of the convergence features in mobile devices equipped with numerous radio interfaces. In line with this trend of making Internet applications run on mobile terminals, the mobile operator 3 in Sweden recently also launched an offer where the MSN Messenger client has been installed and preconfigured in a handset [26], this is just one example of a trend for more and more Internet services to be available via mobile terminals this is enabled because there is less of a gap between the capabilities of a fixed computer and mobile devices with regard to support for the standard Internet functions such as XHTML, Java, and Adobe (Macromedia) Flash, while at the same time both independent service providers and mobile operators are launching more and more internet applications which they have ported to mobile devices. An example is the Agile Mobile tool, which is the equivalent of web-based ICQ, GoogleTalk, AOL, and Messenger combined[27]. Another is the community platform developed by Kenet Works[28], which supplies mobile operators with a youth-focused networking application platform.

Notably Kenet Works in Sweden has contracts with the MVNO Halebop Mobile to supply them with a platform for building and maintaining communities of friends within their network [28]. The value of a network is often said to be proportional to the square of its user-base (Metcalf’s Law [29]), or if we should quote a more thorough analysis; as n log (n) [74]. Hence we can understand the motivation and logic in building large communities as they can potentially generate large revenues for an operator, and increasingly so if the users are persuaded to adopt new behaviours for communicating with each other, e.g. MMSs or SMSs. Services which are highly community-dependent typically include group communications, such as chat or blogs, picture albums, etc. These latter forms of expressing ones self for the enjoyment of your friends have originated in part from the Internet sphere, so it is only natural that native Internet portals, such as the Swedish site Playahead.com have partnered with mobile platform providers like Kenet Works, to ensure their stake in this area. This is an example where the community ecosystem involves three players: the operator’s provisioning of the community service via a portal (Halebop GoGo), the platform provider (in this case Kenet Works), and the service provider (in this case Playahead) to achieve mobility for its user-base. Another approach was taken by Lunarstorm (a competitor of Playahead) who provides a mobile version of its community site to the end-users via a WAP-site (wap.lunarstorm.se).

These are some of the converging services which are beginning to attract followers building on an already established behaviour, ported to the mobile sphere. Extending applications to the mobile world is a highly complex task nevertheless, and, as was mentioned before, the sluggish uptake of WAP, marketed as the “mobile Internet”, was a factor that taught the telecommunications sphere that reinventing well-known phenomena could be both adventurous and difficult. A way to enable the smooth migration of traditional Internet applications could be to preconfigure the device with all settings, as in the case of MSN Mobile in 3’s LG U890 [26], and thus it is simply a question of entering your username to log in to the community. This should be simple for most computer-literate users. Otherwise, providing on-line support to assist the customer in downloading the right WAP-settings, as in the case of Lunarstorm mobile, could make it much easier for the end-user to get started.

Another factor which can make mobile versions of applications especially attractive is the device’s inherent location awareness with the help of techniques such as mobile positioning or GPS. Mobile positioning works by estimating the angle and distance from multiple basestations. Outdoors the approximate location of the user when the measurements from multiple basestations are combined can be determined to about 120 meters accuracy [30]. GPS uses signals from satellites to determine the location of a device to about 3-15 meters accuracy [30, 32], while WLAN-based location determination indoors has approximately the same precision. Some mobile operators, like Telia Friend finder [31] use these mechanisms, to detect if one or more of your friends are nearby so that you can invite them to join you for a cup of coffee. This is another example of a community feature, which works by the operator providing you with the location of your friends, (who must all be subscribers to this operator’s network) instead of using a meta-community like Lunarstorm or Playahead.

Meta-communities and overlay applications have some advantages though, such as their well-recognized brand name and user familiarity. Consider Google Maps for Mobile and MSN Messenger for example; these are applications which have quickly drawn attention to themselves by building upon their existing user base. MSN Messenger partly builds on SIP, so one important prerequisite for launching it in the mobile devices has been platform support for the SIP protocol stack, which according to [22] will be enabled in high-end devices by 2006, so there is clearly support for more Internet applications coming in the new mobile devices being launched this year.

The Internet applications that are most likely to be successful in the convergence area, will be those that take into consideration the limited computer-literacy of the end-user, which build on an already established behaviour, and those that have an attractive GUI which simplifies use by the end-user. As an example of how big an impact preconfiguration of devices, i.e. taking into account the limited abilities of the users, can have on uptake of services, consider that a Southern European operator experienced an increase in the percentage of users from 0,5% to 34% for GPRS and from 1,5% to 31% for MMS after devices were preconfigured with data accounts while support calls related to settings decreased from 33% to a mere 5% of all support calls [22]. Thus simplifying configuration parameters can have a substantial effect on end-user uptake of- and satisfaction with added value services for the end user.

3.2. The digital music market

Music download via the Internet has been around for quite a long time, but it is only recently that we are seeing the emergence of more and more online stores for the legal purchase of content. According to IFPI [33], the number of online music stores has increased from 50 two years ago, to 335 in 2006, but still digital sales only account for about 6% of the industry's total revenues. However, online and mobile music download have recently become the industry's fastest growing revenue channels and could prove to be of increasing importance for content providers. It could have been the lack of interest from the music industry in the early days of file-sharing which led to such a slow evolution of legal music download; the fact that record companies have been reluctant to sell single tracks for downloading at first, only offering content at prices that few customers were willing to pay; but changes in attitudes, the possibility to purchase single tracks and new price points seem to have changed the market.

There were also complicated DRM issues concerning providing music content such as realtones, and ringtones (the latter being distinguished from the former due to being synthesized audio instead of a real audio file). Depending on whether the song is provided as a ringsignal or music track to the mobile, streamed in real-time, or just provided as a "ringback" tone to the mobile that is played while someone is waiting for you to pick up the phone, also makes a difference. If the song is stored on mobile operator’s servers no forward lock or DRM mechanism is needed, unlike the case when you are downloading it to your device, where such a mechanism is used to prevent you from giving a copy to someone else. Today, through Internet music stores, it is also possible to license songs for download that can be played or copied only a limited number of times, so that you can play it on several of your own devices, or burn it to media a certain number of times. These solutions have yet to be perfected, and today it can be difficult for an end-user, to know how the rights objects, without which the music file is unusable, should be managed in order to transfer files between devices. Another way of providing downloaded music content to each device is to require that the user download it to each device, which is a solution that has been employed both by 3 and T-Mobile in Germany. This could be viewed an example of another way in which convergence is promoted instead of hindered, when the downloading is independent of the content that is downloaded and subsequently charged for.

Another approach has been taken by the Swedish mobile operator 3: their music-interested customer downloads a Java application, the “3 Player“, with which the customer can access a list of songs which are pre-selected by the user at the operator’s Internet-portal, for playing on the mobile phone. Initially the price model was that the customer paid only for the price of the upstream data connection of commands to the server, later it changed to mandate subscribing to a music package. This customer offering is similar to offers around the world such as “Napster To Go”, “Yahoo Music Unlimited”, and “Rhapsody To Go”; each one having as the common feature that customers can listen to entire music libraries while on the move. In this way, a customer of 3 can access about half a million songs while not needing to worry about complicated rights issues, which are left entirely up to the mobile operator to handle with the copyright holder[22].

Ultimately, it is important to tailor the applications to suit a particular customer’s needs. Thus, the 3 Player also offers customers the possibility to buy songs if they wish, while 3 OnAir gives the customer the possibility to buy a specific song for playback as a shared media stream, where others might also influence the choice of content being played, and thus a customer can listen to another person’s choice of music as well as their own. With all of these alternatives 3 has succeeded in creating (controlled) virtual spaces where the distribution of content is favourable for both the content owner and the consumer of content, and in this way promoting the provisioning of content via the mobile, while ultimately also contributing to the diversity of music service offerings in the market.

To further enhance a music offering, of course, the mobile device also needs to be equipped with the necessary capabilities in terms of built-in music player support; many devices have a mobile version of Microsoft Window’s Media Player built in to them, others, such as Motorola or Sony have chosen their own music players, or used the well-known Apple Computer Itunes [33] functionality. This is a factor which in the long run hinders convergence, as the different file formats are proprietary to specific vendors such as AAC (proprietary to iTunes), WMA (proprietary to Windows Media Player), or ATRAC3 (proprietary to Sony’s devices), thus hindering convertibility from one format to another. Clearly, this trend hinders convergence in the service offering, and limits the end-user’s ability to transfer and playback songs on many different devices.

With regard to the above, it is evident that music download or streaming of music in various forms could constitute an important factor in promoting convergence between the networks and the devices used for the purpose of playback of the content, if compatibility between devices is ensured. However, at present, business models are quite different and the gap between spreading of illegal versus buying legally acquired content is so great that these constitute obstacles to this development instead of the opposite. Thus, the mobile music market is that is an area where at present differentiated offerings promote divergence in an area where convergence could be enhanced if file-formats were made compatible (or at least convertible) with one another. Paradoxically, it has been the mp3. format which offers no built-in security, that has promoted basic convergence of file-formats between devices in this area, thus far.

3.3. Mobile TV - DVB-H or MBMS?

Another service, which is highlighting convergence from a slightly different perspective is the much anticipated Mobile TV launch. As of this fall in Sweden the state-owned digital television broadcaster, Teracom[35] has started trials of DVB-H broadcasting. DVB-H uses the portions 174 -230 MHz and 470 - 830 MHz of the radio spectrum, and is fundamentally a digital broadcasting technique optimized for use with handheld, battery-powered receivers. DVB-H allows for high data rate in the downlink (from the central mast to the receiver) using a timeslot containing a maximum of 2MB of information. This information is then stored in a buffer at the receiver for later consumption or live playback. The battery power consumption of the device depends on the relation between the on/off time of the receiver. Thus it is more battery-consuming to watch live streaming of a television program, than to play previously stored episodes. Ultimately, DVB-H's success depends on successfully delivering programming that users are willing to pay for. Thus, the IP Datacast Specifications[45] for service discovery, selection, and data protection provided via the telecommunications network is an integral part of the DVB-H specification.

Multimedia Broadcast Service (MBMS) is a standard that has been developed by 3GPP, with the specification adopted in Release 6 in 2004. This version is expected to be adopted by the handset and infrastructure manufacturers leading to devices at the end of 2007 [36]. It allows users to enjoy multicast flows of up to 256 Kbps in UTRAN access networks, and up to 128 Kbps in GERAN networks (if 4 time-slots are used) in either unicast or multicast mode, dependent on an advanced counting scheme which decides upon the optimal use of radio resources within a cell. The user service part of the network can also monitor the successful download of a file to the end user offering FEC or file-repair methods, so that the quality of the content downloaded to the device can be ensured. This is one aspect where MBMS has a significant advantage over DVB-H because it is not a broadcast only technique. Additionally MBMS has quality assurance mechanisms in the downlink. Another benefit with MBMS compared with digital broadcasting techniques is that users can use an uplink channel for interaction with the service. On the other hand, field-trials are already underway with DVB-H, so in this respect it has a competitive advantage over MBMS. However, DVB-H relies upon the buildout of a new DVB network, which can't be coupled with the already existing DVB network, because of the inferior range of the DVB-H antennas, whereas MBMS utilizes either the GSM network or UMTS network, something which could be advantageous.

With Korean mobile television subscriber penetration reaching as high as 30% in 2004, it seems that Mobile TV, although an area of great promise to the mobile operators [22] remains an area where divergence is the norm. In the case of DVB-H, once one manufacturer started delivering devices with support for this technology, other manufacturers feel compelled to do the same in order to maintain market share. Unfortunately, to avoid giving a competitor an advantage in time-to-market many devices represent non-standard implementations, as manufacturing started before the emergence of a global standard. However, if the user experiences a poorer quality of service than expected this may lead to an initial negative first impression of the service, thus creating a barrier for subsequent uptake, something which can be costly, and in the long run doesn't favour convergence in this area.

3.4. Multiplayer Gaming

There is at present a clear trend in the community for game consoles to have online connectivity built-in to the device. Notably the Sony PlayStation Portable was launched with WLAN-compatibility, and mobile devices are increasingly equipped with functionality enabling group communication while maintaining a gaming session with a group of friends; such as presence, push-to-talk, etc. This is an indication that we are moving in a direction of convergence also with regard to gaming offerings. Another example is Microsoft’s recent launch of a unified game platform called “Live Anywhere”, which enables communication between a number of devices of different sizes and shapes playing the same game. So, as one game manufacturer says [22] “…around the corner we see more cross-platform integration between console and PC…”, it is apparent that the gaming industry is focused on enabling gaming anywhere, in any form, not only through the stand-alone time-killers that still have a significant market share[22].

However, manufacturers face a tough task when porting games to devices with different capabilities in terms of battery life, screen resolution, and differing levels of Java-support, (despite the slogan: “write once - run anywhere”). The reality that game manufacturers are faced with is that there exist a multitude of Java Specification Requests (JSR’s) that need to be supported for the optimal performance of a Java game. The Java version that is most widespread for mobile devices is the J2ME; however, mobile devices typically have limited memory (today less than 512Kbytes) so there also exists CLDC and standardized (subset) agreements, such as MIDP, which both have widespread operator support. There exists also an umbrella specification called JSR 185, which is commonly known as the Java Specifications for the Wireless Industry (JTWI) which gives roadmaps for supported functionality in mobile devices to make device functionality coherent over time, which includes a number of different JSR’s for such things as Wireless messaging, Web services, Mobile 3D Graphics etc. The degree to which this specification is adopted is thus a good indication of the coherence of a device‘s functionality with other devices.

These standards mentioned discussed above define subsets of the core APIs as delivered in mobile devices. A feature which increases interoperability between devices is that MIDP version 2 is backwards compatible with MIDP version 1, so any application that has been developed on MIDP 1 will run on a phone which employs version 2 or later. The current release of MIDP is at version 3.0., and this version maintains the aim of ensuring interoperability across devices, so one could argue that these mechanisms are ensuring that the whole community of Java developers can build standardized software that should work on almost any device. Furthermore, OMA Group list Management[48], OMA Presence[48], and JSR 281[48] have also emerged as important enablers for multiplayer gaming. JSR 281 basically includes support for OMA Presence and Group List Management in a Java-application, thus enabling real-time communication across a gaming session. This JSR is followed by several leading chipset and device manufacturers [37, 38]. If manufacturers and game developers adhere to these standards, the chances for interoperability and thus convergence between games that can be played simultaneously on any one device having network access are much greater. However, there is some additional complexity in this picture, as most device manufacturers and many game platform vendors have developed their own additions to Java to optimize their particular devices.

Additionally, for mobile gaming to take off it remains to be proven if the inherent limitations and boundaries of the networks to which one attaches enable a gaming session be kept intact, or if the connectivity breaches imposed by roaming to another network disturbs the ongoing session. Moreover, the Java application needs to support the use of presence and messaging standards independent of access technology and interface utilized, in order for this to be a viable scenario.

All in all, mobile gaming is a complex issue, where a number of factors, such as delay and throughput combine to produce a gaming experience that should to the maximum extent possible be kept ongoing across multiple devices despite differing levels of network connectivity. So, not only do we seem to be faced with the problem of tailoring applications to take into account the different characteristics of the different mobile terminals, but also to exploit the characteristics of the underlying networks in a resourceful way. Therefore it seems logical that some predict an evolution from turn-based multiplayer games to real-time multiplayer sessions, independent of the device or underlying access technology used [22]. Nevertheless, this seems to be an area where there is a consistent movement toward convergence between devices in the future, to be able to keep up with the demand for gaming sessions in numerous devices and via various points of network attachment.

4. Test of VoIP clients in various roaming scenarios

4.1. Motivation for roaming tests of VoIP clients

After discussing the drivers for convergence in turn above, it seemed that the massive attention and uptake that VoIP has seen, driven by clients such as Skype[49] and becasue the different wireless access networks seemed to support the necessary QoS. that Voice over WiFi (VoWiFi) [47] was the most significant driver for convergence. Therefore, with the help of prof. Mark T. Smith, I conducted some sample VoWiFi tests. Originally, the plan was to test both intra-BSS handover in the university's WLAN network, and handover and performance in a public 3G network. However, when I was about to start the tests with the Skype Mobile client, on a Qtek 8310 equipped with both types of interfaces, it was found that the Skype client didn't exist for the terminal in question. So, while investigating the existence of an alternative client, supporting a vertical handover (i.e. a handover between networks of different scale), I ran some tests of intra-BSS handover on an HP iPAQ H5550 and a Fujitsu-Siemens Amilo K7600 series Laptop, using Skype on both machines. All of the following tests were designed to be qualitative in nature rather than quantitative, and the main objective was to studying how the inter-BSS handover and vertical handovers affected the perceived voice performance of the client in the mobile client. Later, in my research I also found an alternative VoIP provider, Woize[51] which provide a client running on the Qtek 8310 that supposedly performs vertical handover of a VoWifi session to GPRS; this was subsequently tested. The results of these tests are discussed in the following sections.

4.2. Definition of VoWifi

The term VoWiFi denotes Voice over WiFi, and is a buoyant area in the market at the moment. There are more and more suppliers, such as Qtek or Sony Ericsson, who could support VoWiFi clients in their new multi-interfaced new terminals [73], this promises to be a very interesting future market. Communications providers such as Optimobile [46] offer an integrated user GUI for terminals that have multiple network interfaces to take advantage of the fastest, cheapest connectivity at any particular location at a given time. Thus, VoIP sessions can be both initiated and maintained within either an IP network or a GSM/3G network, ultimately and ideally leading to a decrease in number of devices associated with each end-user, and an ability to maintain existing communications sessions while roaming between areas of different types of network connectivity. These user clients will likely utilize SIP for their communications sessions. In order to use one of the phones supplied by a communications provider such as Opticall's UniPhone client [47] that supports roaming between different areas of network connectivity, you need to have a subscription with a SIP telephony provider. In Sweden there aren't many Sip providers that are independent of an access network operator at the moment; digisip.se [52] and wx3.se [53] are the only suppliers I have found so far (even though the supplier needn't necessarily be located in Sweden, as it only handles the call signalling). Furthermore there is an issue of whether the different broadband carriers or mobile carriers used for transporting the SIP-based telephony use packet inspection to disallowing SIP packets from entering or originating in their network. At least one of the major broadband providers in Sweden employs packet inspection and thus disallows packets with a SIP header, making it impossible to place calls while using their network, as they think that VoWifi could potentially be a threat to their own telephony solutions [54]. Thus, it is safe to say that such an operator will probably attempt to force the use of their own solutions instead, solutions that aren’t convergence oriented, at least not as of today.

4.2.1. Horizontal handovers’ effect on VoWifi performance

The first VoWiFi testing that I conducted used the following configuration of the two machines running Skype clients:

Configuration of Skype clients:	MN (Hp iPAQ5550)	CN (stationary Fujitsu Siemens laptop)
Processor	Intel X Scale 400 MHz	mobile AMD Athlon XP-M 2800 2.12 GHz
Memory	128 MB SDRAM + 48 MB Flash ROM	480 MB RAM
Operating System	Microsoft Windows Pocket PC 2002	Microsoft Windows XP Home
Memory slot	SD slot: SD, SDIO, and MMC support	-
Network interfaces	WLAN 802.11b, bluetooth	WLAN 802.11b, Ethernet
Battery time with network interface up	2,5h	unlimited (in the case of wired)
Weight	206,5g	2800 g
Audio	Microphone, speaker and headphone jack	Speakers, microphone and headphone jack

Table 1- Configuration of Skype VoIP clients in first test

4.2.1.1 Configuration of Skype Mobile Client on HP iPAQ 5550

In order to conduct the first set of tests, the Skype Mobile client needed to be set up on the HP iPAQ 5550. This was accomplished through the use of Microsoft’s Active Sync® application for synchronization between a mobile device and a desktop/laptop Windows PC. With the help of this mechanism, the CAB (Cabinet) files needed for Skype Mobile to work on the PDA were downloaded to the iPAQ, and installed. Then a connection to a wireless LAN, the KTH open network[58] in the test environment, “Ingenjörshuset” was established. This required both wireless users to establishsecurity credentials to log in to the WLAN as registered users before receiving permission to use the network. Then with the Skype client running and logged in on both the mobile device and the laptop, we began the first set of tests.

4.2.1.2 Test 1: Horizontal handover impact on VoWiFi performance

The first set of tests was conducted in the “Ingenjörshuset” in Kista, an office building equipped with WLAN access points on each floor except for the ground floor. The first tests were conducted with me using the MN and Mark Smith using the laptop as the CN talking via Skype; while the CN was at a fixed location on the 6th floor, the MN was moving from the sixth floor down to the ground floor with the Skype Mobile client running. In order to get some idea of the reception conditions in the building, we checked the configuration of the wireless network in the building for the location of the WLAN access points, to be able to predict when we would lose connectivity [56]. Additionally, I captured the UDP packets which my Skype client was sending and receiving from the Skype supernode in Tele2’s network [55] to see how many packets were transmitted/received by my Skype client. I ran the program IP Interceptor [57] to collect this network layer information. It should be mentioned that as the tests were conducted during summer holidays, we were possibly the only active users of the WLAN in the building. The first problem we encountered during these tests was that the MN’s Skype program quit unexpectedly while the MN was moving from the 5th to the 4th floor in a staircase in the building. The reason for this was a little bit unclear, but it might have been due to the MN experiencing a disconnection from the network. So, the tests were retried, and this time the MN only quit the Skype application on the ground floor, when it was experiencing limited connectivity. Figure 6 shows throughput to and from the CN's Skype application confirming that when no packets had been recieved for ~15s (the last red lines in the figure, each column in the graph representing the throughput during a 5 second interval), then the application quit at the MN's side (although Mark could still hear me after I lost his speech).

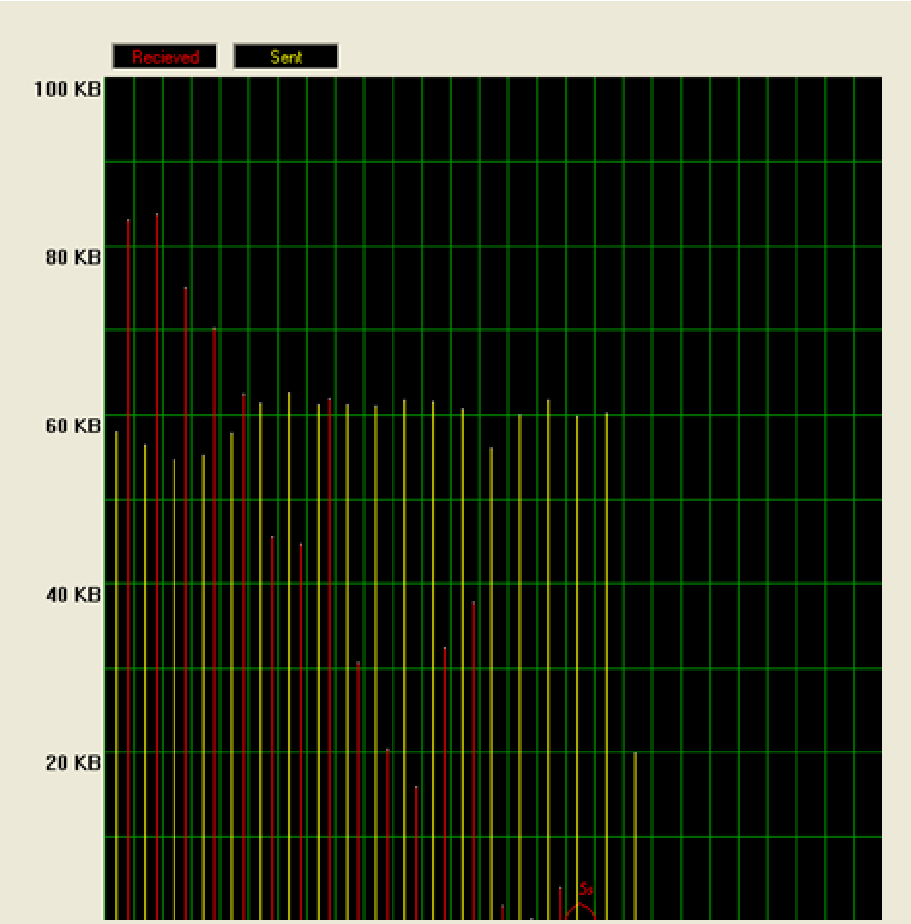

Figure 7 - Ip Interceptor statistics from test 1.

4.2.2. Vertical handover's effect on VoWifi

The second client used in my testing of VoWifi telephony was the Woize™ client, supported by the Qtek 8310. This client was used in the second test concerning vertical handover from VoWiFi to VoIP over GPRS. The configuration of the two devices was as follows:

Configuration of VoIP clients:	MN (Qtek 8310 running Woize)	CN (Fujitsu-Siemens Laptop running Skype)
Processor	TI OMAP 850, 200 MHz	Mobile AMD Athlon, 2.12 GHz
Memory	Flash ROM 64MB, RAM 64 MB DDR	480 MB RAM
Operating System	Microsoft Windows Mobile 5.0	Microsoft Windows XP
Memory slot	Mini SD RAM	-
Network Interfaces	WLAN, GPRS, IrDA, Bluetooth	Wlan 802.11b, Ethernet
Battery time with GPRS up	-	-
Battery time with WLAN up	-	unlimited (in the case of wired)
Weight	106g	2800g
Audio	Speakers, headphone and microphone jack	Speakers, microphone and headphone jack

Table 2 - Configuration of VoIP clients in second test

4.2.2.1. Configuration of Woize Mobile client on Qtek 8310

Configuring Woize on the Qtek 8310 proved to be a little more challenging than configuring the Skype Mobile client on the iPAQ. The first thing needed for testing the effect of sending traffic over GPRS to work was to get a SIM card for a public GPRS network; as the Qtek 8310 only supports GPRS/EDGE packet switched data connections, and not WCDMA. This was arranged for and the GPRS settings were subsequently downloaded to the phone. Active Sync® was once again used to transfer the application and settings for the Woize smartphone application to the phone from the Laptop. The Microsoft .NET Compact Framework, version 2.0 was also needed for the Woize client. When all this was accomplished the WLAN access was established and the second test began.

4.2.2.2. Test 2: Effects of vertical handovers effect on VoWiFi performance

In this scenario, the MN was initially connected to the WLAN, with the Woize client running. I moved from the 6^{th} floor in “Ingengörshuset” to the ground floor, where we expected to loose connectivity. Prior to beginning the tests, we set the Internet settings so that the Qtek would connect to the WLAN as the preferred method to access the Internet. Once this was done, the trials began. Initially the voice quality via the WLAN was good, however after the MN started to move further away from the Access Point it was using, as I moved downwards on the staircase the session quality decreased as the received signal strength decreased. At times, the MN’s speech was not getting through to the CN even though the MN could hear the CN all the time. This test enabled us to examine the handover performance between areas of overlapping/adjacent network coverage, as the WLAN was overlapped by a public GPRS network which could be accessed via the GPRS radio interface, and this network could be used via the subscription information contained in the SIM card inserted in the terminal. The plan was that when WLAN connectivity was lost that the user would switch over to GPRS as the preferred method to access the Internet, while the application was still up and running. However, the Woize client eventually terminated the call when signal quality from the WLAN was such that a certain number of its packets failed to get through (see figure 6). Moreover, the Qtek also switched preferred access method to access the Internet from WLAN to Automatic, when the WLAN network was no longer in range. This indicated clearly that we had lost connectivity and that the Woize application quit due to this fact. However, we didn’t try setting the preferred method to access the Internet to automatic, already when in the building, when we had access to the WLAN, because we presupposed that it would then automatically select the GPRS link as the preferential method to send packets. However, this could be tried out at my home network, where I have an unprotected WLAN connected to the public Internet. This test is described next.

4.2.2.3. Second set of tests with the Qtek 8310

In my home network, I could monitor the WLAN activity with the Woize client up and running, through looking at the activity lights on my WLAN router, knowing that the smartphone was the only device on this network (by looking at the setup page on the router which was simultaneously being accessed by the laptop connected to it via Ethernet). Therefore any activity on the WLAN during the Woize conversation must be due to it being connected to the WLAN. I set the preferred Intenet connection to automatic and started the Woize client to see whether the Woize client selected WLAN or GPRS as its preferred access network for the Woize conversation, with the laptop connected via Ethernet to the LAN. What I could see was that the WLAN activity LED was blinking while I was talking on the smartphone, so subsequently it must mean that WLAN has been set to a higher preference in the phone than GPRS. Then when I was about to perform the vertical handover, I turned off the WLAN, and waited for the GPRS session to pick up. What happened was that the Woize client noticed the disconnection and informed me of this fact, although it **didn't** end the call. Thus I tried to go into the "connections" menu again to set the preferred connection to the Internet to GPRS to see if that would cause it to pick up the call, but the call was still "silent" at both ends although it was still ongoing in both VoIP clients. Most probably the reason for the failure to re-establish the call was that we weren't using Mobile IP or such a network protocol which would have made our change of network address transparent to the application. Eventually, the Skype client was the VoIP client that hung up first, and this led me to believe that the packets from the Woize client in fact weren't getting through to the Internet, which also later proved to be the case. I confirmed this by selecting GPRS as the preferential access method to access the Internet, and then turned off the WLAN before starting up the Woize client again. At this point, I couldn't even manage to log in to the client due to "no connection to the Internet", which was clearly not the case as I could connect to a random webpage via this connection, and experienced no problems. So, either the GPRS operator wasn't allowing the Woize packets to get through and that would explain why the call couldn't be re-established in the first scenario, and why the Woize didn't even start in the second scenario, or that it was inherent in the application that it didn't invoke the right network parameters to set up a GPRS connection. However, what is noticeable is that the Woize client didn't terminate the call despite periods of several seconds of no packets getting through. This is means that if the GPRS connection could have been set up, we probably would have been able to continue the call, and the only reason that we couldn't was that for one reason or other the Woize client couldn't get access to the Internet with the GPRS network I was using. Another reason for the failure to start the Woize application might have been that the MN was behind a NAPT-gateway, and thus the information the application needed to provide to the network was not valid, as the IP address might have been a private one, and hence not publicly routable. This question was raised at the phpBB forum that exists at the Woize website [68], however there were no clues to be found there in this respect.

5. Mobile device components impact with respect to convergence

For the different convergence applications discussed above to run smoothly, there are components that must be well-tuned and protocols that must be well integrated with the software in order to support different types of applications. This section will discuss what the requirements of the different convergence services are in terms of components of the mobile phone. These components can, for the sake of clarity be divided into hardware components and software components. They will be discussed here in light of which impact they might have on convergence. Of course, here there exists a tradeoff of the capabilities needed by an application to run smoothly, and the time before a substantial number of phones with these capabilities are shipped. The important capabilities that affect which applications can be executed on a mobile device include processor power, memory, JSR support, media player version, and Operating System (OS). These features will be discussed in turn in sections 5.1, 5.2, 5.3, and 5.4 respectively. To conclude the section on components capabilities relating to convergence, a table showing the current status of these capabilities in recently launched phones is presented (see Section 5.5.)

5.1. Hardware components: Processor capacity in handsets

One interesting, and most vital component of a mobile terminal is its processor. Processing power is something which, with the increase in use of standard computer components in mobile terminals, is becoming increasingly fast and decreasingly costly. In the course of the two years 2003/2004-2005/2006 processing power in UMTS/GPRS/GSM chipsets doubled in clock rate [22]. Processing capacity in the mobile devices is used for a variety of different things. The main function of the Digital Signal Processors (DSP's) for example is to handle the different radio protocols, but these can also be used for example to speed up MP3 playback, which is often taken care of by the CPU. Other functions of the processors include running the TCP/IP stack and the rendering of web pages. Because this determines how fast the terminal can send and receive packets, the processor performance is a determining factor when it comes to support for VoWifi. This can also be seen in the list of requirements found both on the homepages of both Skype and Woize [61] in their list of requirements for running their smartphone clients. For example, if processor speed is below the required 200MHz required by the Woize client, their client isn't guaranteed to run smoothly on the device. This makes processor speed an important factor when considering the number of devices among the most sold handsets in Sweden that are equipped with the capabilities to handle VoWifi. Processor power is also a key factor when it comes to enabling playout of video streams for applications such as DVB-H or real-time gaming. Today, a 200MHz processor encodes an MPEG4 frames in double the time used by a 200MHz processor together with an assisting dedicated graphics processor, such as NVDIA GoForce 4800 for Mobile [65] for example. Such a graphics processor can support 30 frames per second for comparison the frame rate at the movies is 24fps) thus processors that enable playback of video and audio streams are already now on the market, and have been shipped with phones since end 2005 according to [64]. This should be enable support for rates ranging from 12.5fps to 25fps that have been used with recent DVB-H trials[65] (additionally, in these trials support for the H.264 CODEC was required, and these capabilities in new handsets will be examined later in this section). However, as many manufacturers consider the figures on processor speed of their platforms confidential, this capability had to be left out when presenting an overview of the capacities of the recently launched mobile phones at the end of this section.

5.2. Hardware components: Memory requirements

Memory in the mobile devices has a role when it comes to storing the CAB-files that are needed to load the VoWifi clients we tried on the devices. Woize for the Qtek 8310 needs about ~400Kb of memory space, while the Skype version I tried for the iPAQ 5550 needs a full 5Mb. Additionally, running the program of course also imposes additional run-time memory requirements on the devices. Memory is also a determining factor when it comes to battery consumption during playback of audio; mp3's or other types of formats. We can see from Huang [62] that energy consumption for server playback of an audio file of normal size (~5Mb) using Windows Media Player consumes twice the power on an iPAQ 5550 as compared to playback from local storage. Thus, for music applications to take maximal advantage of the device's features, it is not necessarily best to have the playback buffer at the last entity before the mobile device, although for devices with a limited amount of memory, this may be the only solution that can be adopted. Thus, in order to enable convergence applications such as music download to an increasing number of devices, the amount of memory inherent in the device, thus to a large extent determines which solutions can/will be adopted.

5.3. Software components: Operating system

When it comes to which operating system the mobile device support, a high-end phone (i.e. smartphones and the likes) are typically equipped with some open operating system like Symbian, Windows CE, Palm Source, or RIM. However the BOM (Bill of Materials) for these operating systems is as much as tenfold that of a vendor specific GUI, thus most low-tier phones are equipped with proprietary solutions, making it more difficult for developers to produce applications that will run smoothly on top of such platforms [22]. Typically gaming companies and other application developers will have to port their games to a variety of different devices with differing functionality in terms of networking, JSR support, and messaging, which makes it more complicated to produce streamlined convergence applications that will run on top of any device. This trend is however slowly following the development in the PC business with more and more devices switching from vendor-specific OSs to open operating systems [22]. For many areas of application development such as games, instant messaging solutions, etc. such a development would make it easier to develop generic applications that can successfully take advantage of the device's inherent capabilities, such as networking etc. if they could be accessed via a common, well known API. The top ten selling devices in the summer months in Sweden 2006 will also be classified in this respect in the graphic presented below.

5.4. Software components: Media Player

Media Player audio format compatibility in the mobile devices is fundamental for music convergence offerings to be successful. As of now, many different music players exist on the market, such as Windows Media Player (used mainly in Nokia and Samsung terminals), Sony's media player (used in Sony Ericsson mobiles), Itunes (used by Motorola and in all Apple iPOD's) and these players generally support different formats such as WMA, ATRAC, and AAC; however, many of them also support the mp3 format and this is an important driver for convergence. Online music stores provide music in many different formats, such as WMA (DVD.se) or AAC (Itunes); and it if you don't posses a tool for conversion of the audio files, it may or may not be possible to play your purchase on your specific choice of device. In the next section, I will investigate which of the top selling mobiles that are able to play specific formats.

5.5. Classifying the highest selling mobile device models with respect to convergence

With the objective above of determining whether the most popular devices that are shipped in Sweden as of today have convergence possibilities or not, the table below presents a compilation of the most sold handsets from two out of the four large Swedish operators: Telia and Tre for the summer months June to August 2006. The devices are classified as enabled or not in terms of the capacities discussed above. Finding exact figures of the number of units sold by the major suppliers in Sweden during the summer months of June, July, and August proved to be a difficult task. Consulting statistics released on the Internet from the major mobile telephony providers in Sweden, only gave a summary, calling the manager for retailing for one major operator the figures were considered confidential. Thus I went into the stores and asked salespeople if any phones were preconfigured and capable of VoIP: the answers I got varied from unawareness to denial. This shows that there is some resistance to VoWifi from the mobile operators, since it might compete with their current business. Furthermore, there is the issue of packet inspection, which is being employed by some broadband carriers, rejecting SIP packets, in order to favour their own telephony solutions. Should this trend spread to more operators, this will of course be an important hurdle for VoWifi to surpass. However, convergence services aren't restricted to VoWifi only. Given the definition of convergence services in the introduction (see Section 1) as any solution which builds on the concept of either two or more network architectures coming together, or services migrating from one sphere to another. Devices should preferably have an open operating system that supports at least the basic JSR's that enable applications to take advantage of the underlying network's capabilities. The device also needs to support a number of different CODECs for music/video playout/streaming along with high performance graphics for gaming and music video streaming. Considering this summer's top selling handsets, in the light of some of these aspects of convergence discussed above a classification was made a classification is one which takes into account the accumulated position of the terminal during the three summer months in these two operator's markets is presented. If a mobile handset was number one in the statistics, for one mobile operator three months in a row, it was given the score 10 x 3 = 30. This is only a simple overview of the different features of the handsets that were available, but nevertheless, it shows how convergence-ready they are. A simple qualification scheme for which capabilities the mobiles need to possess are:

- To support VoWifi the device must have support for SIP.
- It needs to be equipped with a fast enough processor to enable rendering of both broadband audio (VoWifi) and streaming video (mobile TV) or real-time gaming.
- It needs to support a common media format: mp3 (audio) or mpeg4 (video).
- There needs to be devices that aren't high-end devices, for the mass-market adoption (VoWifi), so all devices priced over 250 EURO are classified as non mass-market.
- It needs to have at least enough memory to download and install a simple client, or a Java application (for gaming and VoWifi).
- For mobile TV, the number of frames per second and the display size are important features. Mobile TV runs best on devices that have a screen size of at least 2,1 inches [68]

Manufacturer	Nokia **6280** (47p)	Sony Ericsson **K800i** (43p)	Sony Ericsson **W800i** (29p)	Sony Ericsson **Z530i** (23p)	Sony Ericsson **K608i** (21p)	LG **U890** (21p)	Sony Ericsson **K750i** (20p)	Sony Ericsson **K610** (18p)	Nokia **5140** (18p)
Network interfaces	WCDMA 3-band GSM,WLAN	WCDMA 3-band GSM	EDGE 3-band GSM	3-band GSM	WCDMA 3-band GSM	WCDMA 3-band GSM	3-band	WCDMA 3-band	EDGE 3-band
SIP support	Yes (in new software release)	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A
MIDP version	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0
screen size	2,2 inches	2 inches	1,8 inches	N/A (clamshell phone)	1,8 inches	2,2 inches	1,77 inches	1,9 inches	5 graphic rows
Audio codecs supported	Mp3, AAC, AAC+, AMR, m4a	Mp3, AAC, AAC +, m4a, wav	Mp3, AAC, m4a, wav	Mp3, AAC, m4a, wav	Mp3, AAC, wav	Mp3, AAC, AAC+	Mp3, AAC	Mp3, AAC, wav	Mp3
Video codecs supported	3gpp, H.263, mpeg4	3gpp, mpeg4	3gpp, mpeg4, H.263	Mpeg4, 3gpp	mpeg4, 3gpp, H.263	mpeg4	3gpp, mpeg4	mpeg4	H.263
Memory	6MB internal memory + Mini SD slot	64MB internal memory + Memory Stick Micro (M2)	34 MB internal memory + Memory Stick Duo/Pro Duo	28 MB internal memory + Memory Stick Micro (M2)	32MB internal memory	80 MB internal memory + Micro SD/trans flash	34 MB internal memory + Memory Stick Duo/Pro Duo	16 MB internal memory + Memory Stick Micro (M2)	3,5 MB internal memory
Price (EUR) (convesion rate: 1EUR = 9.26SEK)	313 : High-end	411: High- end	313: High-end	146 : Low- end	338 : High-end	346 : High-end	227 : Low-end	259 : High- end	141 : Low- end

Table 3- Top selling handsets classified according to different convergence features

6. Conclusions

What can be seen from the results above is that when it comes to those convergence services that are being introduced in the market, the multitude of services that exist today are enabled in varying degree in the most popular devices. As can be seen from the table on the previous page most prerequisites for interoperability between services in the top-selling phones are not pervasive (with the exemption of gaming). Thus, we see different rates of introduction to the market, and differing obstacles depending on the type of convergence service we are referring to.

VoWiFi is special in the respect as it has seen such a large buzz in the Internet world, and is now looking to conquer the mobile sphere, thus far without any significant effect. One reason for this might be, looking at the table on the previous page, that only high-end phones such as the Nokia 62870 offer WLAN and SIP support. The paradox here is that while phone calls can be made over a WiFi data connection more cheaply, the device is in the high-price segment, meaning that there will be a longer amount of time before a return on investment is achieved. For example if you buy a low-end WLAN box plus a broadband subscription from one of the major operators in order to replace your calls, you would pay ~333kr/month only for this. However, it could be argued that this could replace your Internet connection and thus shouldn't be considered. Fair enough, but take the terminating costs when calling, via a VoIP provider, such as Woize, in average 300 minutes per months (say 50% fixed and 50% mobile) which is the average for Western Europe [75]. Then the cost only for these terminating charges would be greater than the cost of calling via the GSM/WCDMA interface, where you would pay only 0,67 SEK per minute with a subscription that enables you to call an extra 100 minutes a month [76]. Thus there seems to be no benefit in replacing ordinary country wide phone calls with VoWifi calls, at least not for private customers[1]. For the inexperienced computer user, there is also the additional fuss in terms of installing software, learning the GUI, etc. something which many users prefer to avoid. Then there is the issue of whether the mobile operator will attempt to block SIP-packets sent over GPRS rendering such applications futile in any case, something which was stipulated as a possibility in section 4. So clearly, VoWifi, while it represents an area of great promise, one of the obstacles it still faces is that it has to be accepted by the major operators, and they are reluctant to accept competing solutions to their telephony offers, as they now are forced to lower tariffs due to regulatory measures taken by for example the European Union[69]. Furthermore, new communications providers, e.g. Opticall [47] have to pay terminating fees when a call is made to a "regular" mobile telephony user, the difference between using the mobile operator's solution and the VoWiFi provider's solution might in the worst case mean that you only achieve limited economic benefit in using VoWiFi, while loosing the benefit of being able to call your friends while on the move, as there is not yet complete WiFi coverage in urban areas, although this is a trend that is increasing[72]. Thus far the users that have traded their regular mobile phone calls for VoWiFi calls and benefited from this have mainly been users with sufficient computer-literacy that can achieve benefits by saving on long-distance calls, particularly international calls. While this is one of the advantages of VoWiFi, the primary drawback can be the broadband- and mobile operator's potential use of packet inspection and similar measures to block the types of traffic they don't want to terminate/originate in their network. Another is, of course, the limited number of devices that support SIP and WLAN. Some models now being launched have these capabilities

[1] . For business customers this may be different as their company often accepts the acquisition charges for the phone, and thus generating a stock of high-end phones that are used also for "private" phone calls

such as the E60, E61 and E70 by Nokia or Sony Ericsson's P990, but these are still high-end phones, and mass market is thus expected to be one or two more generations away for the consumer segment, meaning roughly in two years time these phones will be availiable.

What can be said about VoWifi is that there is a slight difference between the consumer market and the business market in terms of the business case for VoWifi. If an IT manager at a larger company should decide to acquire terminal equipment and replace their branch exchange by a Wireless Branch Exchange (WBX) that makes use of the company's existing WLAN infrastructure for example, through for example solutions like the one provided by Optimobile [46], the business could potentially save on costs for long-distance calling and because of this, it is an immediate alternative worth considering.
If we then turn our attention to convergence in content offerings, we can see from the table above, that for music download, the evolution is clearly heading in a direction of convergence; all of the top selling mobile terminals (even the low-price segment ones such as the Nokia 5140) have support for the ubiquitous mp3. audio format. This should be seen as an area where convergence has been implemented at a basic level; there still remains the issues of compatibility and transferability of DRM-protected content between a user's different devices, something which unless addressed might foster unlawful file-sharing and spreading of pirated content instead. Thus users could benefit from better alternatives to the forward-lock, which is the most common protection method used in this content segment today [22].

Next, when discussing mobile TV or streaming video, the limiting factors are the screen size, where only 2 of the devices amongst the top 10 selling handsets were equipped with a large enough screen to make watching video an enjoyable experience, with regard to the standard of 2,1 inches of frame size that was set in the focus group session I participated in [67]. Then, there is also the issue of how many frames per second the device can support, however 2 or 3 of the devices tried with video playback in the session are amongst the ones in the table supporting video playback without significant distortions in the picture. However, none of the above devices, support H.264, which was used in recent DVB-H trials[65]. Hence we shouldn't expect television broadcasting to be available for the mass-market soon. Live video streaming using the H.263 together with the AMR audio CODEC is already possible, and can then be converted to 3gp or mp4 streams which according to the table on page 34 would be playable on most of the popular devices today[70]. So, what can be said is that streaming video or television is an area where we are moving slowly toward convergence in the sense of devices which can play either streamed video or stored content, this is largely due to their large memory capacity (note also that only the Sony Ericsson K608 and the Nokia 5140 weren't equipped with expandable memory).

Turning to multiplayer gaming, we see a remarkable correlation amongst all the best-selling models in the Swedish market. All were equipped with MIDP version 2.0. which means they all support Over The Air Provisioning (OTA) of Java midilets, and that they use the same APIs for messaging, gaining access to the Internet, etc. This should at one hand be seen as a significant advantage for the game manufacturers who wish to see an as large as possible stock of interoperable mobile terminals as possible, to avoid having to port the games to many different models (see section 3). Unfortunately, due to various screen sizes some adaptation of the games needs to be done nevertheless. However, the evolution in this area is nevertheless taking a very slow pace, when aiming all the time to ensure backward compatibility which we exemplify summarize by saying that those handsets that are available at this point in time are interoperable with the same underlying MIDP structure (which of course should make the distribution of multiplayer games much easier for manufacturers). This might be something which considerably slows down uptake of games that use the newer features of the devices, and this is sometimes seen as a potential bottleneck for c mobile gaming to take off. However, the fact that different models might perform different actions at different speeds on the device, is something which shouldn’t be that significant for turn-player multiplayer games, but will become important with the introduction of real-time multiplayer games, and can then be expected to be something that stimulates the vendors to optimize their devices for example to reduce latency, to tailor the device for a specific audience. I would propose therefore that games manufacturers aim to fully utilize new features of the handsets, such as location determination, and multiple interfaces to keep up with the evolution of the devices, while continuing to make the gaming experience consistent across a number of devices, for convergence to be promoted in the most successful way.

Finally, briefly to comment on the Instant Messaging and community solutions that are available in the market today, such as the ones from Microsoft, Kenet Works, and Playahead, it seems that these solutions, which as already stated in section 3 aim to increase the activity of the communities of users in the network are the ones that will be most closely linked to the operators, and will probably follow an evolutionary path which is tightly coupled with the path that the operator is pursuing. Thus, these applications can be considered drivers for convergence, but to a lesser extent than the others mentioned above, as the mobile operators will promote them on their own and may subsequently choose either a slow pace or a fast pace in introducing them to the market.

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Appendix A - Abbreviations and definitions.

AP	Access Point
ARQ	Automatic Repeat Request
API	Application Programming Interface
BSS	Basic Service Set
CLDC	Connected Limited Device Configuration
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CS	Circuit Switched
DIFS	Distributed Interframe Spacing
DHCP	Dynamic Host Configuration Protocol
DoS	Denial of Service
DRM	Digital Rights Management
DSP	Digital Signal Processing/or
DVB-H	Digital Video Broadcast for Handhelds
EAP-AKA	Extensible Authentication Protocol- UMTS Authentication and Key Agreement (using USIM)
EAP-SIM	Extensible Authentication Protocol- Subscriber Identity Module
E-GPRS	Enhanced GPRS
ENUM	Telephone Number Mapping
ETSI	European Telecommunications Standardizations Institute
FA	Foreign Agent
GAN	Generic Access Network
GANC	Generic Access Network Controller
GERAN	GSM EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GTP	GPRS Tunneling Protocol
HA	Home Agent
HLR	Home Location Register
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IKE	Internet Key Exchange
IMS	IP Multimedia Subsystem
IM	Instant Messaging
IPSec	IP Security
ITU-T	International Telecommunications Union, Telecommunications Standards Sector
ISUP	ISDN User Part
J2ME	Java 2 Micro Edition
LLC	Link Layer Control
MAC	Medium Access Control
MAP	Mobile Application Part
MBMS	Multimedia Broadcast Multicast Service
MIDP	Mobile Information Device Profile
MN	Mobile Node
MSISDN	Mobile Station International ISDN Number
MVNO	Mobile Virtual Network Operator

NAPT	Network Address Port Translation
OMA	Open Mobile Alliance
OSI	Open Systems Interconnection
OTA	Over The Air provisioning
PBX	Private Branch Exchange
PDP	Packet Data Protocol
PLMN	Public Land Mobile Networks
PSTN	Public Switched Telephony Networks
RFC	Internet Request For Comment
RLC	Radio Link Control
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SIPS	Secure Session Initiation Protocol
SMS	Small messaging service
SNR	Signal-to-Noise Ratio
SRTP	Secure Real Time Protocol
TCAP	Transaction Capabilities Application Part
TCP	Transmission Control Protocol
3GPP	Third Generation Partnership Project
TLS	Transport Layer Security
UDP	User Datagram Protocol
UMA	Unlicensed Mobile Access
URI	Uniform Resource Identifier
UTRAN	UMTS Terrestrial Radio Access Network
UMTS	Universal Mobile Telecommunications System
WAP	Wireless Application Protocol
WAN	Wide Area Network
WCDMA	Wideband Code Division Multiple Access
WEP	Wireless Equivalent Privacy
WiFi	Wireless Fidelity
WLAN	Wireless Local Area Network
VL	Visitor Location
VoIP	Voice Over IP
VoWiFi	Voice over Wifi
WPA	Wireless Protected Access

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