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Mobile VOIP

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Preface

Our agenda with this assignment is to look into various wireless technologies, IP telephony and various approaches of converging these. We have chosen to limit ourselves to write an overview of the different FMC solutions and their techniques. Our goal is to discuss the differences between the FMC solutions, to get an overview of how far they are from getting realized and a market survey.

What we hope to learn from this is an insight into a new technology that is right around the corner. We wish to get an introduction of the techniques that are proposed for these solutions and to compare these. We also wish to conduct a market survey to better understand how the market would respond to the introduction of mobile VoIP.

We want to thank our teacher Jo Herstad and Telio, one of the main VoIP service providers in Norway, that gave us the opportunity to look into this field and for guiding us on the right track.

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Document structure

To make it easier for the reader, here is a short outline of the document:
Part 1: Background information and technology (Chap 1 and 2)
Part 2: Fixed mobile convergence, solutions from various vendors and comparison(Chap 3 and 4)
Part 3: Context aware communication and market survey (Chap 5 and 6)
Part 4: Conclusion of the previous parts(Chap 7)

Contents

Parameter	Specification
`Multiple access`	TDMA/FDMA
`Duplex`	FDD
`Channel bandwidth`	200 kHz
`Uplink band`	890-915 MHz
`Downlink band`	935-960 MHz
`Forward/reverse channel data rate`	270 kbps/user
`Number if users/channel`	8

Table 1: GSM Spesifikasjonen

1 Background technologies

In this section we will look in to the underlying wireless network technologies needed to be able to set up a mobile VOIP solution.

1.1 Cellular networks

A cellular network is a radio network made up of a number of radio cells. The essential elements of any cellular system are mobile stations (or mobile unit), base stations, and a mobile switching center.

There are multiple cellular network standardizations throughout the world. The cellular system which is in use in Europe uses the GSM standard.

1.1.1 GSM

GSM, Global System for Mobile Communication, is a second-generation cellular system. Some of the specifications of GSM are shown in table 1

GSM has led to a more integrated mobile network because each operator supports GSM-based infrastructure elements, GSM standard formats, and GSM handsets.

1.1.2 GPRS

GPRS, General Packet Radio Service, is a GSM data transmission technique that does not set up a continuous channel from a portable terminal for the transmission and reception of data, but transmits and receives data in packets, which means that multiple users share the same transmission channel, only transmitting when they have data to send.

The total available bandwidth is dedicated to those users who are sending at any time. GPRS deliver effective data rates up to 50 Kbps.

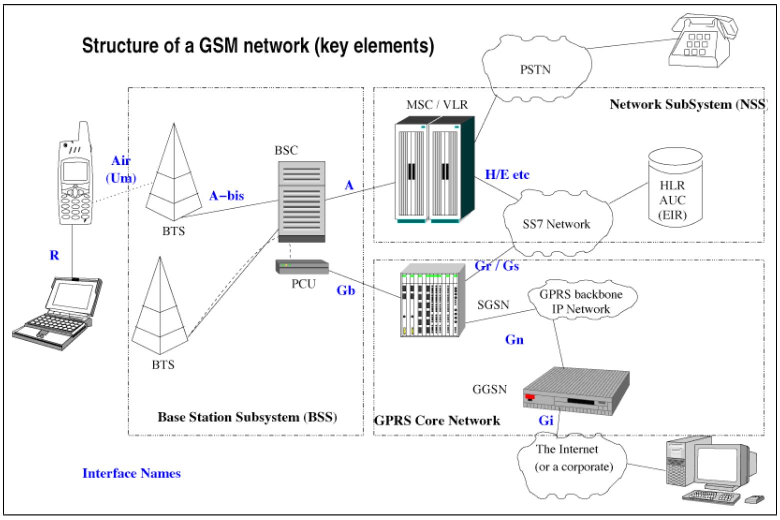

Figure 1: GSM

1.1.3 3G

3G, Third Generation mobile technology, supports much higher data rates, measured in Mbps and there for provide the ability to transfer live video, audio and full Internet access.

3G is a IMT-2000 specification made by International Telecommunications Union.s and supposed to be a single, unified, worldwide standard, but in practice, the 3G world has been split into various camps.

For the 3G to be compatible with GSM, the Europeans adopted the UMTS standard which is based on CDMA technology

1.2 Wireless LAN (WLAN)

WLAN stands for wireless local area network. It is based on the 802.11 family of specifications developed by IEEE for wireless LAN technology. 802.11 specifies an over the air interface between a wireless client and a base station or between two wireless clients. WLAN cover areas with a radius of up to 100 meters and it is optimized for indoor range. There are several specifications in the 802.11 family:

1.2.1 802.11

Applies to wireless LANs and provides 1 or 2 Mbps transmission in the 2.4 GHz band using either frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (DSSS).

1.2.2 802.11a

Is an extension to 802.11 that applies to wireless LANs and provides up to 54 Mbps in the 5GHz band. 802.11a uses an Orthogonal Frequency Division Multiplexing (OFDM) encoding scheme, to support higher data rates rather than FHSS or DSSS.
Products that adhere to this standard are considered Wi-Fi Certified. It is less potential for RF interference than 802.11b and 802.11g. Better than 802.11b at supporting multimedia voice, video and large-image applications in densely populated user environments, but it has relatively shorter range than 802.11b.

1.2.3 802.11b

Is an extension to 802.11 that applies to wireless LANS and provides 11 Mbps transmission (with a fallback to 5.5, 2 and 1 Mbps) in the 2.4 GHz band. 802.11b uses only DSSS.
Products that adhere to this standard are considered Wi-Fi Certified. Not interoperable with 802.11a. It requires fewer access points than 802.11a for coverage of large areas and offers high-speed access to data at up to 100 meters from base station.

1.2.4 802.11g

Applies to wireless LANs and provides 20+ Mbps in the 2.4 GHz band. Which of these wireless LAN technologies should be employed depends on the purpose and the applications that will be used. Range and performance are the big issues to be considered.

802.11a offers excellent support to higher end applications involving video, voice and transmission of large images and files, but the higher operating frequencies equates to relatively shorter range. RF interference presented in 802.11b is avoided in 802.11a, since it operates in the less crowded 5GHZ band and not in the 2.4 GHZ band which 802.11b operates.

1.3 WiMax

WiMAX stands for. World Interoperability for Microwave Access.. It is a global standard based technology for Broadband Wireless Access that supports fixed, nomadic, portable and mobile access. WiMAX is an IP-based Radio Access System configured in much the same way as a traditional cellular network with strategically located base stations using a point to multi-point architecture based on IEEE and ETSI standards.

In its first release the 802.16 standards addressed applications in licensed bands in the 10 to 66 GHZ frequency range. It was optimal for cells with a 7 to 10 km and up to 50 km of range. Future amendments

Table 1. Types of access to a WiMAX network

Definition	Devices	Locations/ Speed	Handoffs	802.16-2004	802.16e
Fixed access	Outdoor and indoor CPEs	Single/ Stationary	No	Yes	Yes
Nomadic access	Indoor CPEs, PCMCIA cards	Multiple/ Stationary	No	Yes	Yes
Portability	Laptop PCMCIA or mini cards	Multiple/ Walking speed	Hard handoffs	No	Yes
Simple mobility	Laptop PCMCIA or mini cards, PDAs or smartphones	Multiple/ Low vehicular speed	Hard handoffs	No	Yes
Full mobility	Laptop PCMCIA or mini cards, PDAs or smartphones	Multiple/ High vehicular speed	Soft handoffs	No	Yes

Figure 2: WiMax

have extended the 802.16 air interface standards to cover non-line of sight (NLOS) applications in bands in the sub 11 GHZ frequency range and sub 6 GHZ frequency range for mobile WiMAX.

1.3.1 Fixed-WiMAX based on the IEEE 802.16-2004

Fixed WiMAX has proven to be a cost effective fixed wireless alternative to conventional wired-line DSL and cable in areas where those technologies are already available and none less in areas beyond the reach of DSL and cable.

It uses Orthogonal Frequency Division Multiplexing (OFDM) with 256 carriers and supports fixed and nomadic access in Line of Sight (LOS) and Non Line of Sight (NLOS) environments. Using the 3.5 GHZ frequency range licensed band and the 5.8 GHZ frequency range unlicensed band. It has a typical cell radius of between 5 to 8 km.

WiMAX profiles based on 802.16-2004 are better suited to fixed applications that use directional antennae because OFDM is less complex than SOFDMA. As a result 802.16-2004 networks may be deployed faster and products are less complex and can be used in a wider range of unlicensed bands.

1.3.2 Mobile WiMAX based on the IEEE 802.16e

The 802.16e is a amendment add to the 802.16 standard by IEEE, it adds the features and attributes to the standard that are necessary to support mobility.

Mobile WiMAX is a broadband wireless solution that enables conver-

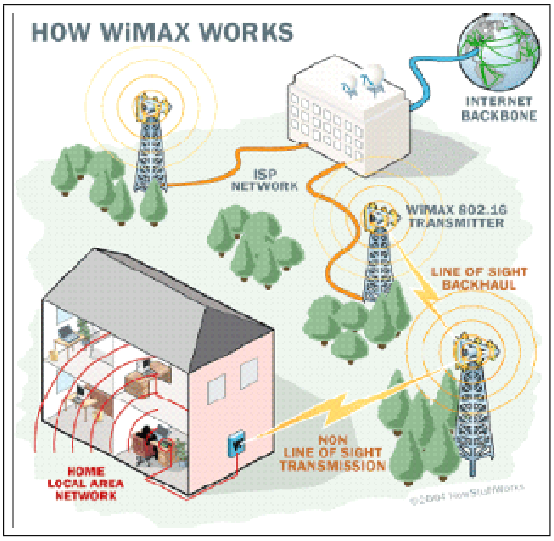

Figure 3: WiMax 802.16-2004

gence of mobile and fixed broadband networks through a common wide area broadband radio access technology and flexible network architecture.

It is optimized for dynamic mobile radio channels and provides support for roaming and handoffs. Handoffs are a crucial ability to maintain a connection while moving across cell borders. 802.16e WiMAX will support both hard and soft handoffs. Hard handoffs use a break-before-make approach, the user device is connected to only one base station. It is less complex than soft handoffs, but has a late latency. Soft handoffs are much alike those used in some cellular networks and allow the user device to retain the connection until it is associated (make-before-break approach), thus reducing latency. Applications like VoIP benefit greatly by low latency soft handoffs. QoS are maintained during handoffs.

It uses Scalable Orthogonal Frequency Division Multiplexing Access (SOFDMA, a variation of OFDMA), a multi-carrier modulation technique that uses sub-channelization. The carrier allocation in OFDMA modes is designed to minimize the effect of the interference on user devices with omni directional antennae. Furthermore the 802.16e offers improved support to Multiple Input Multiple Output (MIMO) and Adaptive Antenna Systems (AAS) which will bring a substantial increase in throughput and NLOS capabilities. Mobile WiMAX has a typical cell radius of 1.5 to 5 km. The frequency bands for the 802.16e profiles are not yet certified, but 2.3 GHZ and 2.5 GHZ are the best targets due to better indoor coverage and support for mobile or portable

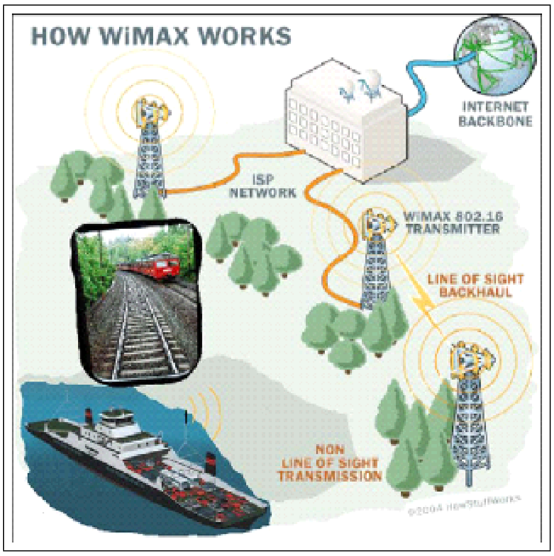

Figure 4: WiMax 802.16e

devices.

Some advantages presented by the 802.16e products compared to the 802.16-2004 products are a better link margin, support for mobility, improved indoor coverage and flexible management of spectrum resources.

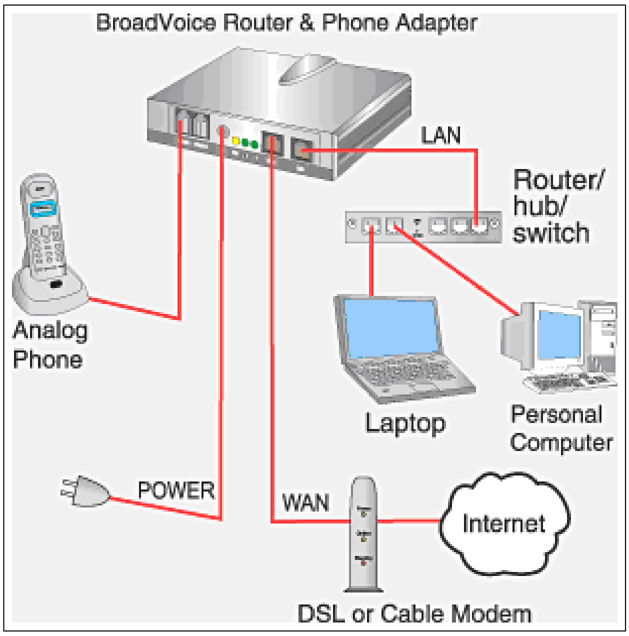

Figure 5: VOIP

2 VOIP

Voice over IP allows you to make phone calls using a computer network, over a data network like the Internet. Voice over IP converts the voice signal from your telephone into a digital signal that travels over the internet then converts it back at the other end so you can speak to anyone with a regular phone number. When placing a voice over IP call using a phone with an adapter, you.ll hear a dial tone and dial just as you always have. Voice over IP may also allow you to make a call directly from a computer using a conventional telephone or a microphone.

At present there are two standards that are in use for VoIP switching and gateways: SIP (Session Initiation Protocol) and H.323. SIP mainly relates to end-user IP Telephony applications, while H.323 is a new ITU standard for routing between the circuit-switched and packet-switched worlds used for termination of an IP originated call on the PSTN, but the converse is also becoming common at a very fast rate.

The most popular of these two standards and most widely used is SIP.

2.1 SIP

SIP clients use TCP and UDP port 5060 to connect to SIP servers. SIP is only used in setting up and tearing down voice or video calls. All voice/video communications are done over RTP. A goal for SIP was to provide a superset of the call processing functions and features present in the public switched telephone network (PSTN). As such, features that permit familiar telephone-like operations are present: dialing a number, causing a phone to ring, hearing ringback tones or a busy signal. Implementation and terminology are different.

2.2 H.323

H.323 was designed with a primary target: To provide teleconference with voice and data capacities on packet switching networks.

The continuous researchs and developments of H.323 follow the same purpose. In addition, H.323 and the convergence of voice, video and data allow the services suppliers to provide new facilities for the users and improve the performance for the user reducing costs.

The standard was designed specifically with the following objectives:

- To be based on the existing standards, including H.320, RTP and Q.931
- To add some of the advantages that packet switching networks offer to transport real time data
- To solve the problems of real time data on packet switching networks

The designers of H.323 know that the requirements of the communication differ from one place to another, between users and between companies and obviously the requirements of future applications also change. So, the designers of H.323 defined it in such a way that the companies that manufacture the equipment can add their own specifications to the protocol and can define other structures and standards that allow the devices to acquire new features or capacities.

2.3 SER

SIP Express Router is a high-performance, configurable, free SIP server. It can act as SIP registrar, proxy or redirect server. Its performance allows it to deal with operational burdens, such as broken network components, attacks, power-up reboots and rapidly growing user population. SER.s configuration ability meets needs of a whole range of scenarios including small office use, enterprise PBX replacements and carrier services.

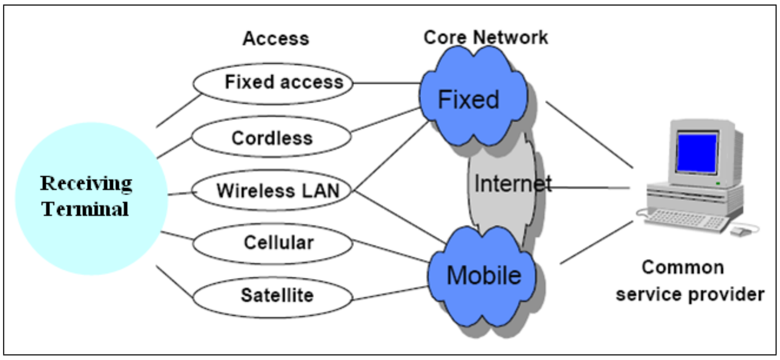

Figure 6: FMC

3 Mobile VOIP

3.1 Fixed Mobile Convergence

Fixed mobile convergence (FMC) is a generic term that embraces terminal device, service and network convergence. That is merging wire-line and wireless networks, service and terminals. With the convergence between the mobile and fixed networks, telecommunications operators can provide services to users independent of their location, access technology, and terminal.

The concept of convergence emerges from telecom service providers need to find new revenue stream, reduce their operating expenses and simultaneously invest in future-proof network architectures and technologies. Some service providers are looking for a multitude of new services including mobile and fixed access. The primary goal is concurrent delivery of all media type (Voice, data and video) to an easy to use graphical user interface, independent of access method, terminal and location. The goal of network convergence is to make all service profitable and enable multiple business models. These goals are related because frequently service that are easy to use become popular and increases revenue. The convergence can then be seen in tree aspects or levels, the core network, terminals and services.

3.1.1 Network convergence

Network convergence means that same network will be used for both fixed and mobile service and by both operators. This part can be farther divided in to core network and access network the goal for the core network is to migrate from separate circuit and packet switched networks to a single unified network that supports the existing mobile and fixed access technology.

3.1.2 Terminal convergence

Terminals convergence is that terminals should be interoperable across multiple access technologies and vendor networks seamlessly.

3.1.3 Service convergence

Service convergence is to be able to provide/access new or existing service in both fixed and mobile network independent of your location. This can be composed of one or combined service, such as videophone. An important future of this is that users can access a consistent set of services from any fixed or mobile terminal via any compatible access point, Independent of access network it is attached to.

3.1.4 FMC trend

The fully fixed mobile converged service and network are same years a way, but there are some attempts that are been made. We have IMS, UMA, H323 and SIP. Session initiation protocol is a protocol developed by the ITFG and proposed standard for initiating, modifying, and terminating an interactive user session that involves multimedia elements. It is gaining popularity compared to the complementary protocol H323.The IP Multimedia Subsystem (IMS) is a standardized Next Generation Networking (NGN) architecture for telecom operators that want to provide mobile and fixed multimedia services. It uses a Voice-over-IP (VoIP) implementation based on a 3GPP standardized implementation of SIP, and runs over the standard Internet Protocol (IP). Existing phone systems (both packet-switched and circuit-switched) are supported.

3.2 IMS

IMS was originally designed for the mobile network by 3G.IP, witch was formed by 1999 defined the original IMS and developed the initial IMS architecture, then it was brought to the 3rd Generation Partnership Project (3GPP), as part of their standardization work for 3G mobile phone systems in UMTS networks. 3GPP2 (a different organization) based their CDMA2000 Multimedia Domain (MMD) on 3GPP IMS and added support for CDMA2000.

IMS is Next Generation Networking (NGN) multimedia service architecture for operators that want to provide mobile and fixed multimedia services. It is an overlay network on the top of internet. Existing phone systems (both packet-switched and circuit-switched) are supported. The aim of IMS is to provide all the services, current and future that the Internet can provide. IMS uses open standard IP protocols (defined by the IETF) to establish all multimedia session between two users (IMS - IMS /IMS - Internet / Internet - Internet) and as interface for service developers.

IMS defines eight core network functions (elements). The elements can be grouped in tree main groups, call control, media processing and gateway.

The Call/Session Control Function (CSCF) handles the call control function. Signaling messages from SIP are processed by the CSCF. The CSCF functions include Proxy CSCF which performs like an SIP User Agent and handles the forwarding of SIP requests and responses. The Interrogating CSCF is also an SIP proxy and functions as a contact point for the operator.s network and the users that are roaming that network; the serving CSCF is an SIP server that helps to maintain the state of a session as desired by the network provider.

The Media Resource Functions (MRF) manage media processing. The Media Resource Function Processor (MRFP) handles functions such as the processing of mixed incoming media streams, audio transcoding, announcements, etc; the Media Resource Function Controller (MRFC) performs a controlling function for the media stream resources within the MRFP.

The gateway consists of the Breakout Gateway Control Function (BWCF) which is an SIP server that performs routing functions to allow calls between networks. This enables calls to begin on the IMS and finish on the PSTN; the Media Gateway Control Function (MGCF) transforms the signaling protocols such that they can be used by a given network; the Media Gateway (MGW) is used for changing the media streams that are used on a network such as RTP to a format that can be used by another network such as the PCM.

3.2.1 IMS Architecture

The IMS architecture is based on tree layer architecture (see figure 7 on the next page), collection of functions and standardized interfaces for linking the functions. The tree layers are Application, IMS core and transport layer. IMS architecture is an overlay architecture witch is run over the Internet.

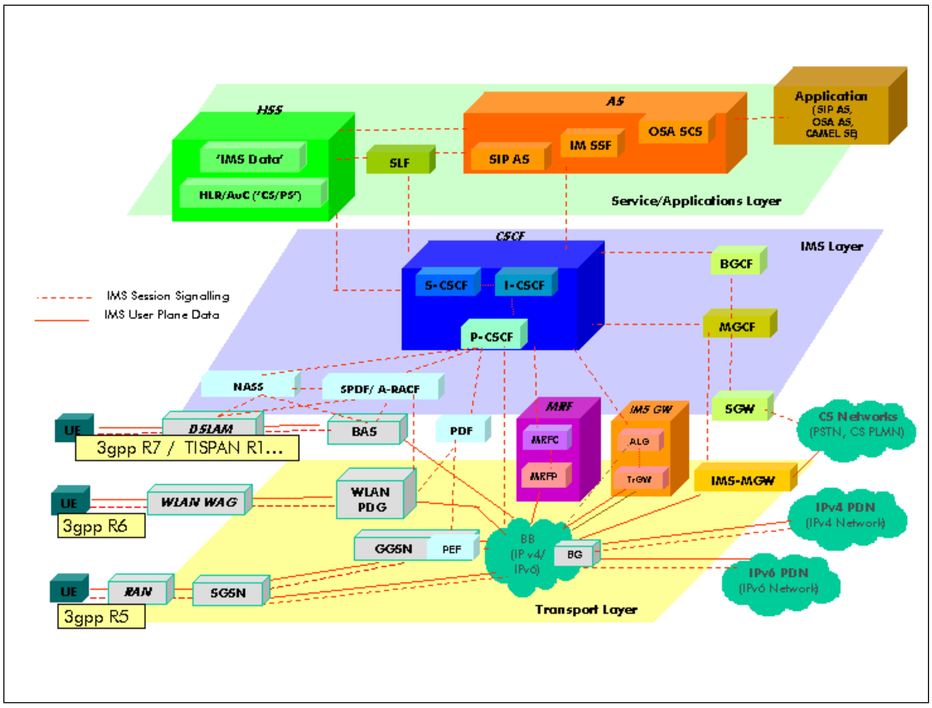

Figure 7: IMS Architecture

4 FMC solutions

In this section we present seven different solutions for IMS and fixed mobile convergence, and compare them to find the solution that is most comliant with IMS.

4.1 UMA

Unlicensed Mobile Access (UMA) technology provides access to GSM and GPRS mobile services over unlicensed spectrum technologies, including Bluetooth and 802.11. By deploying UMA technology, service providers can enable subscribers to roam and handover between cellular networks and public and private unlicensed wireless networks using dual-mode mobile handsets.

4.1.1 How it works

When a subscriber with an UMA enabled handset is in range of an unlicensed wireless network to which the subscriber is allowed to connect, the handset will automatically connect. Upon connection the handset will authenticate with the UNC (UMA Network Controller) over the IP / broadband network. Authentication is required to access GSM voice and GPRS service via the unlicensed wireless network. If the authen-

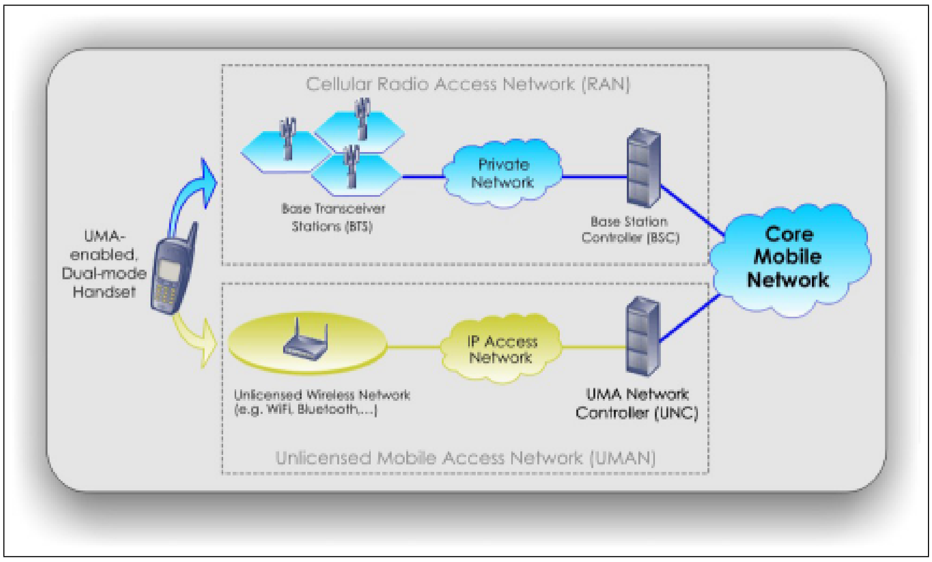

Figure 8: UMA

tication is successful the subscribers location is updated in the core mobile network. Now all voice and data traffic is routed to the subscriber through the unlicensed mobile access network (UMAN). This is outlined in figure 8.

4.1.2 Roaming

When a UMA-enabled subscriber moves outside the range of an unlicensed wireless network to which they are connected, the UNC and handset facilitate roaming back to the licensed outdoor network (GSM / 3G). This roaming process is completely transparent to the subscriber.

4.1.3 Handover

If a subscriber is on an active GSM voice call or GPRS data session when they come within range (or out of range) of an unlicensed wireless network, that voice call or data session can automatically handover between access networks with no discernable service interruption. Handovers are completely transparent to the subscriber.

4.1.4 Benefits

Mobile operators tend not to deploy new technologies unless they provide either a cost benefit or enhanced user experience. UMA technology delivers on both counts by providing improved voice quality and in-building coverage while offloading traffic from existing (higher cost) GSM radio networks.

A normal GSM cell radius in suburban environments is around 2 km. This gives an approximate coverage area of 12.5 km2. Within this cell one could potentially cover more than 12 000 homes, but due to in-building penetration loss the actual coverage can be reduced by up to 70 percent to less than 4000 of the original 12 000 homes. This reduction in radio access and voice quality will ultimately lead to loss of potential revenue for the mobile operator. This is precisely where UMA technology can be a benefit to both operators and end-users.

With a UMA-enabled phone the subscribers will offload traffic on the radio access network (RAN) to local wireless networks. Not only will mobile operators not have to invest in additional base stations to increase coverage, neither will they have to pay for the access points and internet connections as this infrastructure in many cases already is in place and used for internet access.

4.1.5 Is UMA a fixed to mobile convergence (FMC) technology?

UMA is not in total compliance with the specification for fixed to mobile convergence. After all UMA does not in its self provide convergence between the fixed and the mobile networks, it only provides convergence between mobile and wireless networks. But with that said UMA still contains solutions for some of the basic ideas in the FMC specification.

Mobile operators will begin deploying UMA-compliant dual-mode GSM/Wi-Fi handsets with the objective of offering true “one phone, one number” service. Although this initially may be viewed as complementary to existing fixed service, the long-term result could be mobile substitution of fixed service.

From this viewpoint, UMA technology is an extension of FMC with the ultimate goal of mobile substitution in a converged world. A recent survey conducted by BrainJuicer, which targeted 1,000 customers in six European markets (France, Germany, Italy, Spain, Sweden and the United Kingdom), concluded that a UMA-enabled phone would be positively received. If mobile calls in the home were priced the same as fixed line calls, then more than 50 percent of the respondents said they would be likely to sign up for UMA service within 12 months. Of the respondents who would probably buy the service, a third would make most or all of their calls at home on their mobile phone. This makes for a powerful case for mobile substitution.

As initial deployments of UMA handsets begin with the movement toward “one phone, one number” service, the stage will be set for a

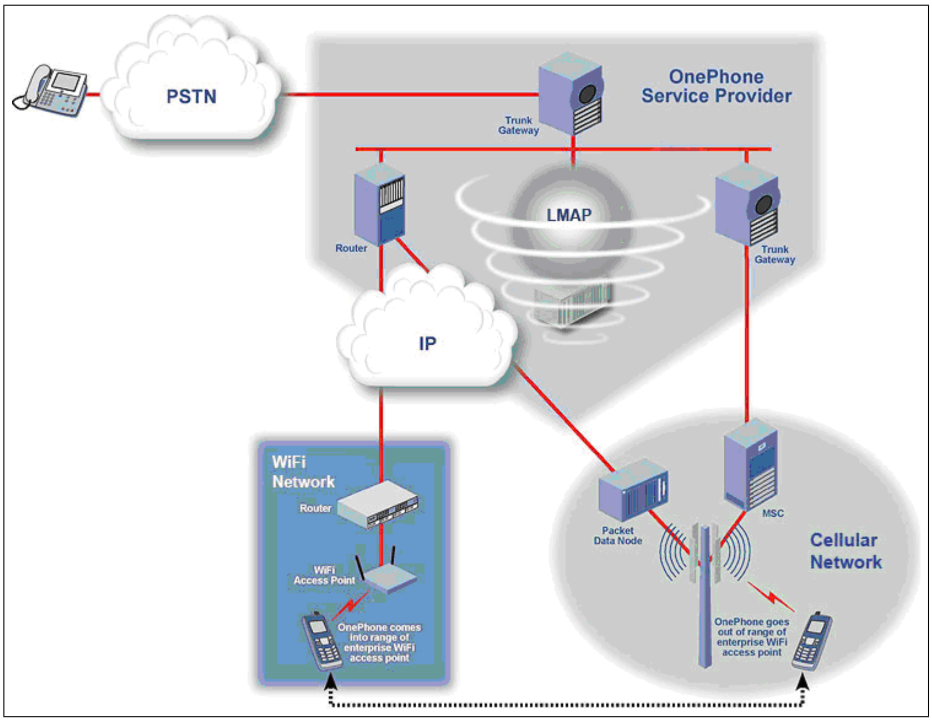

Figure 9: Longboard FMC

more significant movement toward FMC. This also seems to be the case in the marketing of the UMA equipment. Almost all of the technology companies that develop equipment for the UMA backbone argue that deploying UMA is a solution that will increase use of mobile services and effectively stimulate the IP multimedia subsystem (IMS) for mobile.

4.2 Longboard

LongBoard is a software company that specializes in developing revenue-generating, IP-based multimedia and voice applications for leading telecommunications carriers and service providers worldwide.

4.2.1 LMAP

The LongBoard Multimedia Application Platform provides SIP-based session control, with the functionality to serve as a SIP applications server.

At the most basic level, LongBoard Multimedia Application Platform (LMAP) provides a platform for working with SIP user agents (phones / gateways / softclients) to provide line-side features, including basic

Class 5 residential features, hosted business features, and more advanced converged applications, such as the OnePhone application. SIP user agents first register with, and are authenticated by, LMAP. LMAP then acts as an applications server for those endpoints, enabling them with telephony features. When the registered SIP users wish to make calls to the PSTN (or to other SIP users) LMAP can proxy their calls to the appropriate next-hop for delivery to their ultimate destination.

LMAP has been designed to meet the stringent requirements of the world s leading telecommunication service providers for network and component performance and reliability, and has been performance tested by Miercom, an independent test lab. The product sustained over 150 calls per second (compared to a Nortel Class 5 Switch, which sustains 138 calls per second), while achieving 100% uptime, and maintaining a 99.997% call completion rate in 2001.

LongBoards contribution to the FMC is IMS complaint and a SIP client called OnePhone.

4.2.2 OnePhone 2.0

The OnePhone is a personal mobility application. that currently is in trials with leading service providers worldwide. It.s a software that is installed in service providers. networks and in .dual mode. WiFi/cellular phones.

With OnePhone and a dual mode device, business users and consumers can enjoy the cost savings of Voice over WiFi while having the convenience of a single device and a consistent set of services from anywhere . at the office, on the road, or at home.

The two critical capabilities that enable and add value to Fixed-Mobile Convergence:

- First, OnePhone automatically and transparently switches calls between cellular and WiFi networks. This .handover. which occurs in a fraction of a second, enables users to roam seamlessly between cellular and WiFi networks with no interruption in service. OnePhone cleverly senses when a user has entered a WiFi zone, or is in range of a WiFi network, enabling users to save money by reducing usage of expensive cellular minutes.
- Second, and very importantly, OnePhone adds unique value by delivering dozens of highly valuable advanced business features such as call waiting, call transfer, three-way calling, and other features to users of dual mode devices. Future .personal mobility applications. from LongBoard will further enhance the dual mode

phone user experience by adding presence, instant messaging, instant conferencing, and other advanced multimedia services.

The technical benefits for service providers are listed as follows:

- Combat churn and wireline-to-wireless substitution
- Increase average revenue per user (ARPU)
- Capture new customers
- Offer differentiated new .3G. services today while migrating to 3G at your own pace

For the Mobile Users

- Increase mobility and convenience
- Reduce costs
- Increase productivity

4.3 Bridgeport

BridgePort Networks was founded in 2001 and it has its headquartered in Chicago, Illinois. BridgePort-Networks was the founder of MobileIGNITE [10], where it is an active member today and seeks to accelerate the network convergence opportunity. The carrier customer segments for MobileIGNITE include: wireless, wire line, broadband ISPs and Mobile Virtual Networks Operators (MVNOs) and integrated operators who offer services to consumers and enterprises.

The core product of BridgPort-Networks is, the NomadicONE. This is a software based network infrastructure solution. The solution is based on open standard, as Session Initiation Protocol (SIP), SS7 and an open architecture to develop MobileVoIP solution. This makes it interoperable with other third party network element witch are based on open standers. The NomadicONE can be deployed without requiring network-wide upgrades. NomadicONE is run on IBM BladeServer hardware, Linux operating system and Oracle Enterprise Database and it supports Windows mobile 2003 and 2005, Symbian V9 and feature phones.

The NomadicONE is provided in to different options, a pre-IMS NomadicONE (NCG) [6] and NomadicONE IMS convergence server (ICS) [5]. The NCG is not fully incompliant with the IMS specification of 3GPP or 3GPP2, While ICS is fully compliant with bought 3GPP [2] and 3Gpp2. This is done to lower the investment risk by giving a clear evolution path (see figure 10 on page 22). The solution is based in the following platforms.

4.3.1 NomadicONE Network convergence Gateway (NCG)

The NomadicONE Network convergence Gateway (NCG) is a server deployed in the service provider.s network that bridges the mobile and broadband networks, enable users to have one phone number across all device and network, permits SIP based fixed terminals to terminate call and massage made/send to mobile phone number. The NomadicONE NCG can be configured to act as proxy/register server and a Serving Mobile Switching Center (S-MSC)/Visitor Location Registrar (VLR) or MSC Gateway (G-MSC), depending whether it is deployed on mobile or broadband network.

1. G-MSC enables service providers to .extend. existing fixed numbers onto a mobile network. The Gateway MSC also allows consistency of Enterprise services (e.g. extension calling, call forwarding) between SIP-based servers such as IP PBX or IP Centrex and mobile networks

2. As S-MSC it provides the inter working for registration, call processing and handover between the mobile and fixed network.

3. Acts as a presence server between the mobile network and the IP network. Levering presence, the NCG can send/deliver SMS, data, and other multimedia services to a subscriber using the optimal network based on bandwidth, reception, and cost.

4. NomadicONE NCG solution architecture can easily be integrated into the service providers existing network and can operate with a virtually any SIP device, including multimode Mobile + Wi-Fi phones, IP desk phones, IM or PC soft phone clients and fixed-line phones utilizing ATAs (analog telephone adapters).

It supports mobileVoIP handover based on ICM/CTM. Therefore, pre-IMS and IMS dual-mode handsets will be compatible.

4.3.2 NomadicONE IMS Convergence server (ICS)

The NomadicONE IMS Convergence Gateway (ICG) builds on the NomadicONE Network Converged Gateway architecture and integrates new functionality and interfaces required to operate in IMS networks. The NomadicONE ICS will act as application server in IMS network. To communicate with other IMS networks, managing the call legs, determine which network to use and more, the NomadicONE ICS implements all the Call/Session Control Function (CSCF), Network Domain Selection (NeDS) and Home Subscriber Server (HSS). NomadicONE ICS implements as well the 3GPP IMS Control Model (ICM) [2] and the 3GPP2 Call Transfer Model (CTM) [1] for MobileVoIP handover between Wi-Fi and cellular networks[7]. The NomadicONE ICS capability can be achieved by software update to NomadicONE NCG if you have one.

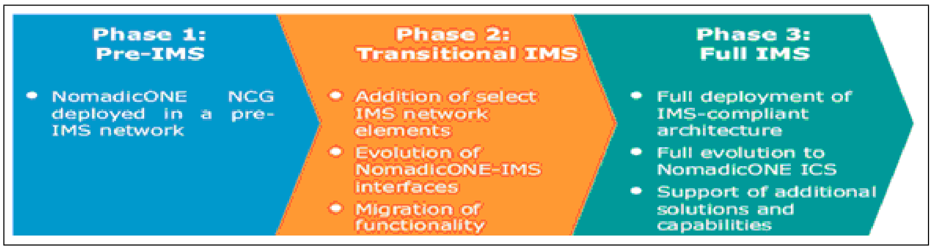

Figure 10: Bridgeport

4.3.3 BridgePort-Network.s Path to all IP

BridgePort-Networks provides a tree phase based path for providers to go from pre IMS to IMS network (se figure 1). The NomadicONE NCN can be developed to go to phase two by selective software update or using third party network element that are based on open standards. It can upgrade the NCG to ICS and other third party network element to go to full IMS. This is outlined in figure 10.

4.4 Siemens

Siemens is one of the world.s largest electrical engineering and electronics companies. Siemens provides innovative technologies and comprehensive know-how to benefit customers in 190 countries. [12]

The Siemens IMS/FMC solution features transparent roaming and active handoffs across fixed and mobile networks. The solution includes personalized multimedia services and for enterprises presence-aware collaboration technologies that help improve workflow and business process efficiencies.

Converged network consumer features of the Siemens IMS/FMC portfolio include:

- Intelligent Address BookStored on the Siemens IMS/FMC subsystem and, therefore, accessible on all networks, contact information may only need to be entered once and is available to all end devices, mobile phones, PDAs, desktop phones and laptops. The application is secured to prohibit other subscribers from accessing it.

- Roaming ButtonAllows a subscriber to receive calls and messages on the device that he or she is currently using. The roaming feature can be enabled or disabled based on presence and availability information.

- Call-and-ShareData and voice work together with this feature. While calling from home to a friend with a mobile phone to explain directions, for example, a subscriber could also sketch a route on a city map and send the instructions to the friends mobile phone during the conversation.

4.4.1 How does the IMS platform works?

In a complex mobile service landscape wherein the operator has deployed a large number of services, it is absolutely crucial that the operator is able to control the invocation of services and the interaction between the various service components [13].

When a user registers on the MNOs IMS network his Subscriber Service Profile (SSP) is downloaded by the Call Session Control Function (CSCF) from the Home Subscriber Service (HSS). The SSP contains a great deal of servicerelated information per individual end user and enables the CSCF to:

- Determine the order in which multiple services are executed (if applicable)
- DDetermine the address(es) of the application server(s)which should execute the requested end user service(s).
- DInform the application server(s) of the order in which services should be executed in the case that multiple services need to be executed on the same application server(s).

4.4.2 Services provided

Dynamic Multimedia Session Control: The circuit switched domain only allows users to have one type of service per bearer or session (and severely limits the session media types).

Multiple Services , Single Session: This allows the user to choose one type of service and ignore rest of the services.

Synchronised Services, Multiple Sessions: This allows the user to use multiple services which are interlinked and can trigger other types of services within discreet, independent sessions.

Unrelated Services, Multiple Sessions: This last scenario involves the user having a number of unrelated services running in parallel, independent sessions.

4.5 Cirpak

CIRPACK, a Thomson business unit, develops next generation Class-5 voice switches for massive deployments of broadband telephony and IP Centrex as well as for migration of legacy PSTN infrastructures to

IMS/TISPAN architectures. CIRPACK deployed the largest SoftSwitch-based PSTN as early as 1998 and is now at the heart of some of Europe s largest VoIP networks.

CIRPACK develops universal voice-switching solutions allowing telecom service providers and carriers to provide innovative value-added services on any types of local loops while consolidating infrastructures and applications according to the IMS model.

CIRPACK SoftSwitch solutions are a full IP Multimedia Subsystem architecture. The Home Subscriber Server (HSS), the central database of an IMS network, and the Call Session Control Function (I-CSCF and S-CSCF), the core components of an IMS architecture, responsible for routing requests from users to actually deliver any multimedia services even when roaming.

The Thomson core IMS solution leverages the CIRPACK softswitch platform, a field-proven Application Server (AS) for telephony and IP Centrex with an IMS/TISPAN Access Gateway Control Function (AGCF) already deployed by over 75 telecom operators in 28 countries on 3 continents servicing over 3 million VoIP subscribers. The CIRPACK platform enables operators to quickly deploy VoIP and SIP-based multimedia services at lower costs. It can be later enhanced by simple software add-on to become a comprehensive IMS core network solution.

The CIRPACK HVS SoftSwitch ensures MGCF and AGCF functions of the IMS/TISPAN reference model. It also provides the Class-5/IP Centrex application server (AS) with its dedicated MRFC/MRFP required to offer voice services. The CIRPACK Media/Signaling gateway ensures SGF and MGF functions of the IMS/TISPAN reference model. All these components used together make a very comprehensive Class-4/Class-5 voice switching solution designed for PSTN Emulation according to the TISPAN specifications.

Enhanced with CIRPACK S-CSCF, I-CSCF and HSS/SLF, the CIRPACK platform becomes a fully compliant IMS solution also capable of PSTN Simulation and managing non-conversational services according to the 3GPP/IMS and ETSI/TISPAN reference models.

Mobile operators using Thomson.s Cirpack IP Centrex platform can start marketing global telephony solutions, offering all the features of hosted PBX solutions, without forcing users to change their mobile handsets. The result is a single service with one phone number and one voicemail box shared by the customer s cellular and broadband

IP phones. Subscribers can enjoy a rich set of features across both devices, including simultaneous ring, call filtering, extension dialling, multi-party conferencing, hunt groups, etc. Cirpack.s Mobile IP Centrex application is currently deployed in trials with several GSM operators in Europe.

Thomson.s Cirpack platform is a highly modular and scalable public telephony switch, incorporating all the software and hardware components required to connect to and interwork with the legacy telephony systems telecom service providers are operating. It has native support for protocols used in GSM infrastructures, enabling seamless integration with existing HLR, MSC and Intelligent Network platforms to ease the introduction of innovative services that also leverage Cirpack.s VoIP and automatic handover of GSM to VoIP.

4.6 Ericsson

Ericsson has an evoulutionary apoach to FMC and IMS. Due to the current state of most PSTN and GSM networks Ericsson introduces an evolutionary model for telecom companies to evolve in to a full IMS solution.

The first step in Ericssons solution is to implement a product called Softswitch. [8]

4.6.1 Softswitch

A Softswitch provides call/session control and media gateway control, in a split architecture based on packet technology separating media/payload and call control/set-up signalling. Softswitching is key in the transition of voice and multimedia communications to a packet-based infrastructure using standard protocols such as (SIP, SIP-T/I, H.323, H.248, BICC, SIGTRAN).

Softswitch solution elements are feature server/application server (e.g. telephony server, mobile server, multimedia server), media server, media gateway, signaling gateway and management, provisioning and charging/billing interfaces.

4.6.2 Implementation

Ericsson has provided to different implementations for FMC, the first is for the currently fixed networks and the second is for current GSM / mobile networks. This seems reasenoble because not all fixed operators operate mobile networks and vice versa.

4.7 Motorola

The Motorola IMS is a carrier-grade system, both GSM and CDMA capable, and based on the highly flexible and scaleable SoftSwitch technology, featuring open standard technology with proven interoperability.

Motorola.s IMS solution offers following services:

- Access Independence
- Converged Push-To-Talk over Cellular (Converged PoC)
- Presence Server
- Unified Messaging
- Interactive Voice Response (IVR)
- Enhanced Voice Mail
- Instant Messaging
- Web/Audio/Video Conferencing
- Full Duplex Video Telephony
- The Motorola IMS Application Partner Program

The Motorola IMS is based on Motorola.s advanced SoftSwitch technology, a flexible, open computing platform that decouples call control from bearer traffic functions. The Motorola IMS is compliant with 3GPP and 3GPP2 standards for IP Multimedia Subsystems, and supports a variety of 2.5G and 3G wireless access networks, including GPRS, EDGE, UMTS, and CDMA, as well as emerging systems such as IEEE 802.11 WiFi, wireline and enterprise networks. It takes advantage of the network independence of IP to create core applications that are agnostic to radio access technologies and that can seamlessly bring new services and applications to wireless networks. Multimedia applications built on the IMS architecture are portable.

Motorola.s IMS can use SIP to establish, modify and terminate multimedia sessions or calls, enabling the protocol to be the basis of applications incorporating voice, video, chat, interactive games, and more. For operators of networks within the GSM airline family (GPRS/EDGE/UMTS), Motorola.s IMS incorporates CAMEL (Customized Application for Mobile network Enhanced Logic), a service creation platform that makes it possible to support worldwide operator specific services. Using standard IN interfaces, CAMEL provides access into the switch platforms, registers and billing systems of the GSM network, making it possible for third parties to create IN services. CAMEL also supports call screening and supervision services, number translation services, enhanced call forwarding (time and location dependent), and fraud information gathering services. Motorola.s IMS also complies with Parlay/OSA specifications for network independent Application Protocol Interfaces (APIs).

APIs standardize and simplify access to core network functionality, allowing developers to concentrate on application quality rather than on the intricacies of network-specific integration.

To further speed and simplify the creation and deployment of new services, operators can turn to Motorola.s Global Applications Management Architecture (GAMA), a seamless management environment that offers easier, more consistent control over every stage of data services development, deployment and operations. Motorola GAMA is both device and network-bearer independent, enabling the development and launch of new services with existing infrastructure and end user devices enabled with JAVA. and WAP. In addition, our IMS Application Partner Program will be an ongoing source of new, tested and proven applications for the Motorola IMS.

4.8 Comparison

4.8.1 UMA vs IMS

Seamless roaming and handover between cellular networks and unlicensed spectrum technologies like Bluetooth, wifi, using a dual mode mobile handset.

UMA has the disadvantage of been only compatible with GSM, unlike IMS that is compatible with other cellular networks such as UMTS. But as a result, it gained an early attraction particulary in places where GSM is the predominant cellular network used. It extends cellular circuit switched signalling protocol by taking the SS7 signalling protocol and tunnelling it to a UMA-end devise. This happens in a nutshell in which all calls stay on the cellular network instead of been converted to VoIP, eliminating cannibalization.

Another disadvantage with UMA is that it is expensive to get UMA base-dual-mode devise in addition to subscribe to an additional service. This rises the question of whether it is a good investment simply to get a better reception.

Something positive about UMA is that it is easier to implement than IMS. This gives the opportunity to deploy it quickly.

UMA was the first FMC solution in the market, but the lack if support for IP-PBX.s and its inability to hand over between multiple access points. makes the future of UMA as a predominant FMC solution uncertain.

On the other hand IMS is based in SIP, which permits convergence. Since it is compatible with UMTS, DSL and other networks, it provides flexibility for developing various solutions for enterprises.

IMS delivers high quality and reliability, things that are expected by the end user. This is possible since IMS is a IP based multimedia and telephony core network, that enables to way voice, data and video across multiples access technologies and devises. Another advantage of IMS is that is uses the same framework for any kind of access (wired or wireless). It supports IP sessions over the internet and simplifies converged applications, utilizing existing and nearly developed applications.

Some benefits for service providers when using IMS solutions are the reduce of costs and investments protection across access technologies, while increasing revenue.

Benefits provided to the end user might be maintaining common contacts across multiple services. It adds multimedia information to enrich communications and collaboration.

4.8.2 Comparing FMC/IMS solutions

Comparing some FMC solutions should give us a better understanding of how to get to FMC by either using IMS or UMA. It might give us an overview of the different implementations this FMC solutions use, and what they take into consideration while developing these products.

LMAP is a SIP application server delivered by Longbord. It provides a platform for working with a SIP user agent. LMAP has already been tested and has proven to be a stable solution that gives reliability, sustaining 150 calls per seconds.

Longbord delivers a software call One Phone 2.0, this software is installed in service provider.s networks and in dual-mode-phones. These solutions delivered by Longbord do not act as a full FMC solution, but it does works like it when combined with an IMS network. Although it does provide automatically call switch between cellular and WiFi networks. On the other hand the FMC solution delivered by Cirpack, Softswitch, is a voice switching solution developed according to the IMS model CIRPACK SoftSwitch solutions are a full IP Multimedia Subsystem architecture. The Home Subscriber Server (HSS), the central database of an IMS network, and the Call Session Control Function

(I-CSCF and S-CSCF), are the core components of an IMS architecture, responsible for routing requests from users to actually deliver any multimedia services even when roaming. The CIRPACK platform enables operators to quickly deploy VoIP and SIP-based multimedia services at lower costs. It can be later enhanced by simple software add-on to become a comprehensive IMS core network solution.

Enhanced with CIRPACK S-CSCF, I-CSCF and HSS/SLF, the CIRPACK platform becomes a fully compliant IMS solution also capable of PSTN Simulation and managing non-conversational services according to the 3GPP/IMS and ETSI/TISPAN reference models. It has native support for protocols used in GSM infrastructures, enabling seamless integration with existing HLR, MSC and Intelligent Network platforms to ease the introduction of innovative services that also leverage Cirpack.s VoIP and automatic handover of GSM to VoIP.

Bridgeport network delivers a software base network called NomadicONE, similar to the software delivered by Longbord. It is based on Session Initiation Protocol (SIP), SS7 wich makes it interoperable with other third party networks.

The NomadicONE is provided in to different options, a pre-IMS NomadicONE (NCG) and NomadicONE IMS convergence server (ICS). The NCG is not fully incompliant with the IMS specification of 3GPP or 3GPP2, While ICS is fully compliant with bought 3GPP [4] and 3Gpp2. The NomadicONE NCG can be configured to act as proxy/register server and a Serving Mobile Switching Center (S-MSC)/Visitor Location Registrar (VLR) or MSC Gateway (G-MSC), depending whether it is deployed on mobile or broadband network. An interesting characteristic with NomadicONE (NCG) is that it can be deployed without requiring network wide upgrade. As well as Cirpack Softswitch solutions do, NomadicONE needs the use of analog telephone adapters (ATA.s) for integrating fixed line phones.

The other option, NomadicONE IMS convergence server (ICS) is build on the NCG architecture. It acts as an application server in IMS, supporting between WiFi and cellular networks.

5 Contextual communications

Communications today has been more or less redefined. Now a days its all about understanding each others standpoints. To understand one another one also has to understand the background of each other. The need for information of each other is apparent. These criterias are becoming more and more fulfilled as information continues to flow free on the internet.

The software developers are creating profiles, where they categorize people by. These profiles make it easier to develop software specifically toward a targeted group of people. Other ways of profiling is to optimize communications. Today all major phone operators of the world are considering ways to converge the fixed network with GSM, resulting in a merging of all sorts of transport whether it be data, sound or video. This is how communications has been redefined, because now you will have more data available to evaluate the information source. You will get images, audio, video, data in close to real time wherever the communication source is in the globe.

5.1 Awareness

"Sharing an awareness of ones environment and the context of friends, family, and colleagues can help determine if another person is available for a communication, and can also create serendipitous opportunities for communication. For example, Instant Messenger uses a status (Online, Away, On the Phone, etc.) that gives an indication of a person.s availability and willingness to chat." [4]

During our research on the different FMC solutions, some of the software available for operators choosing an IMS solution aimed to combine WiFi and GSM on dual mode mobile phones. These software takes aim to gather information on one.s environment and make it available to one.s contacts.

"Future .personal mobility applications. from LongBoard will further enhance the dual mode phone user experience by adding presence, instant messaging, instant conferencing, and other advanced multimedia services." [9]

Dual mode phones that feature the ability to access WiFi and GSM are the next generation phone features. The next step then is to allocate small sensors in public areas where useful information is needed, like bus stops, train stations etc. As soon as the dual mode phone is in contact with the sensors access point, the phone will have access to in example buss tabs, time tables for trains etc.

"Context-aware communication dimensions. Context (e.g, location) can be entered manually or sensed automatically, and the communication act (e.g, call routing) can be achieved manually, with user assistant 1 automatically detects and displays user location, but requires a per-

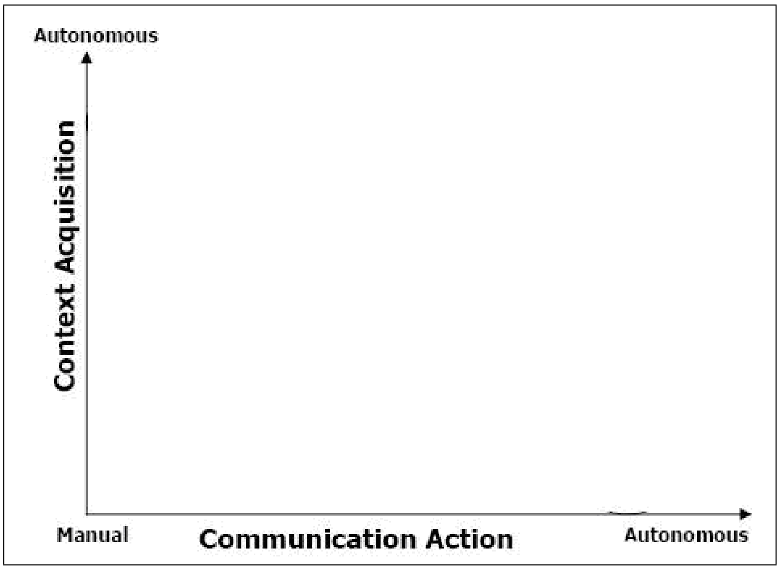

Figure 11: Context aware communication

son to forward telephone calls. “ [4]

SIP [11] based FMC solutions have a Softswitch that routes all of the incoming calls automatically, as for the context, this too moves toward autonomous.

For the average phone caller this means that every time the caller is in range of a WiFi zone, the phone automatically routes all its calls within the internet on a SIP based routing to make it economically and to add more features based on applications made for the SIP protocol. On the other hand one have UMA [3] which also is a FMC solution, but its less autonomous then the SIP based FMC solutions. UMA is in general more stationary, where the cost is not so much cheaper and the caller is limited to be around the router. It is also based on dual mode phones, but ultimately one ends up on the GSM-net.

It seems that toward the future, IMS is what all things tend to boil down to. The routing is autonomous and based on a Softswitch, the services are more integrated. Its based on SIP, which means that data, audio and video are transferable on a economically way. As time requires a more economically and more efficient way to communicate, it seems that technology is not far in its solutions.

One can off course debate the privacy of the IMS technology, as the services are more and more improved.

“As context acquisition becomes more autonomous, recipients may feel their privacy is eroding because systems will become more and more aware of their day-to-day activities. Thus, reducing the work required from recipients may also reduce their sense of privacy. In addition, as communication actions become more autonomous, recipients and initiators alike will notice a reduction in .common sense. as humans are removed from the loop. “ [4]
The statement above has a very interesting truth to it. Because are we not fond of our privacy and anonymity? And as technology are more and more autonomous, don.t we loose our common sense? In the sense that our basic abilities to solve problems will be inhibited because we have to much gadgets that automatically solve our day-to-day problems.

6 Market survey

To get a better understanding on how the market would respond to the introdcution of mobile VOIP we decided to do a market survey. The survey was conducted among fellow students, teacher and colleauges.

6.1 Questions

These are the 9 questions we came up with:

1. Are you answering for your self or as a business?
2. Would you be willing to install a box in your home / business to be able to use mobile IP?
3. What would be most important if you were to use a mobile IP service?
4. Would it be useful to only have one phone number for all telecommunication needs? (Business, private, mobile etc)
5. Would it be useful that you could "log on" to this number from any phone available? (Office phones, mobile phones, hotel phones etc)
6. Would it be useful to see the status of the person you want to call and also be able to set your own status? (Like MSN; Away, Busy, At Work, At Home etc)
7. Would you like you phone to automatically select which network it should use?
8. What type of "sensing service" would be most useful to you?
9. Is there any other services you would like to see when mobile VOIP is introduced?

6.2 Results

In total 141 persons answered our survey. This is a quite high number considering the short period of time the survey was available online. We got the following results:

1. Private: 96,5% - Bussiness: 3,5%
2. Yes: 82,3% - No: 17,7%
3. Price: 89,4% - Quality: 74,5% - Additional Services: 12%
4. Yes: 52,5% - No: 47,5%
5. Yes: 71,6% - No: 28,4%
6. Yes: 84,4% - No: 15,6%
7. Automatic: 73% - Manual: 27%

8. Automatic profile: 61% - Friend finder: 42,5% - Other: 26,2%

A more detailed graphic presentation can be found in Appendix 1.

On question 3 and 8 multiple answers was allowed. Question number 9 was an open question where the users could type in any comment about future services they would like to see, and we got many good sudgestions that we will present bellow under findings.

6.3 Findings

We find that people are generally positive to the introduction of a mobile VOIP service. It is very clear that both price and quality is important for the users if they were to buy a mobile VOIP service. From questions 4 and 5 we find that not all people are interested in having only one number, but they would like the flexibility to be able to log on to their number from any IMS enabled device. People are also enthusiastic about the posibility to set their availability status and also see other users status before making a call. In general people would like their mobilephones to automatically select the best suited network at their current location. When it comes to automatic sensining services people would like their phones to automatically set the profile by "sensing" where they are. Also "sensing" where you friends are seem to be a service people would like to see.

In question 9 we opened opp for sudgestions, and people came up with a lot of good ones. Here are just a few selected:

- Automatically name lookup on incomming call from unknown number.
- The possibility to transfer files to the person you are talking to. In example a picture or videoclip.
- View the call-rate before placing call.
- The ability to have phones ring on different locations.

7 Conclution

The main goal of our project is to find the implementation that adapts the FMC specification best. Through a market research we would like to find out what potential customers want and the potential of an FMC implementation.
Mobile service VoIP is one of the important part in a FMC implementation. Unfortunately, mobile telephony is still based on traditional circuit-switching, and the move to VoIP has barely begun. The keys to operators success are FMC. In the long term, IMS offers mobile operators their best shot at the enterprise FMC market, but it costs. Many mobile operators use for example UMA as an interim solution. But to offer a complete FMC, IMS is a must. There are different solutions on the marked; some of them offer a total insoluble solution which leads to a big investment; some of them offer a part of IMS and has to integrate with other current solutions. In our view the best way to implement a full IMS solution is to implement it in stages. That way the operators will not have to make a large investment at an early stage. The market research views interest for VoIP and other FMC functions as far as the solution is cheap and easy to use.

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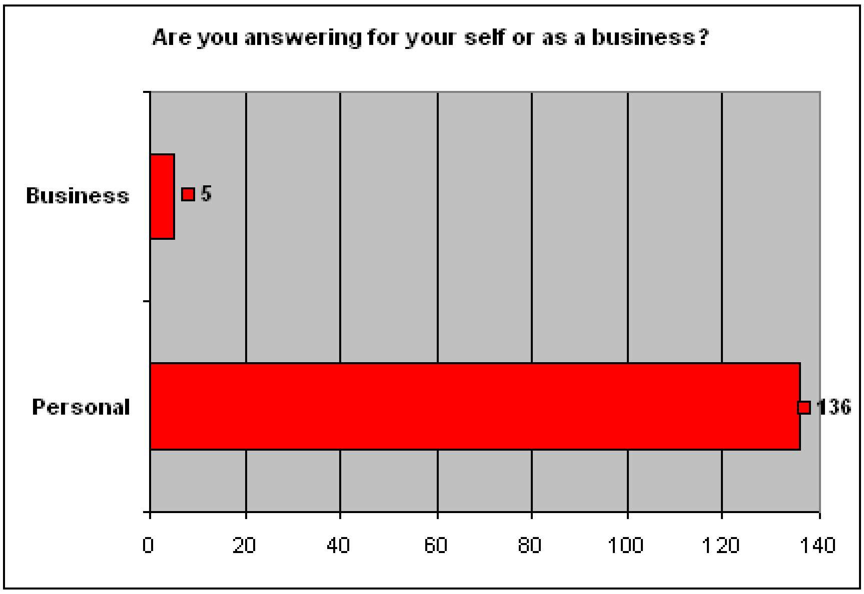
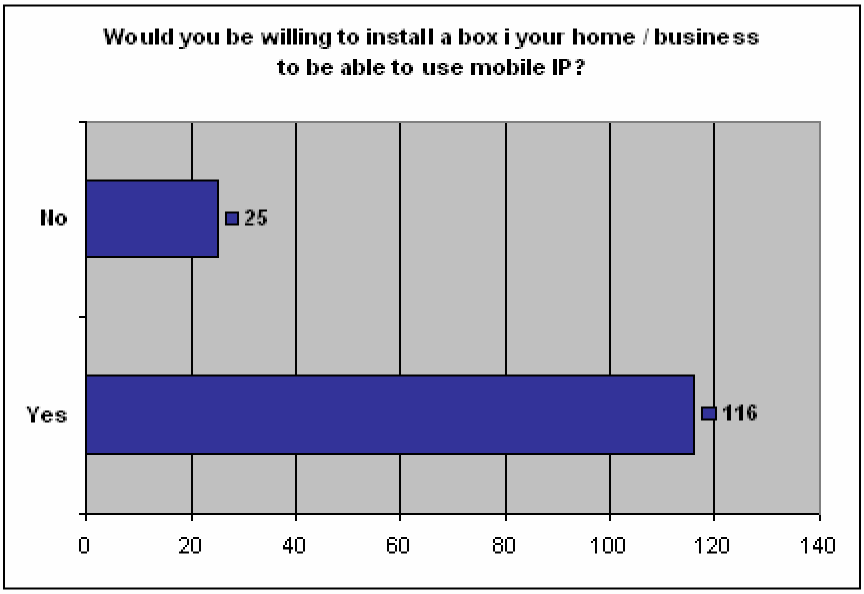
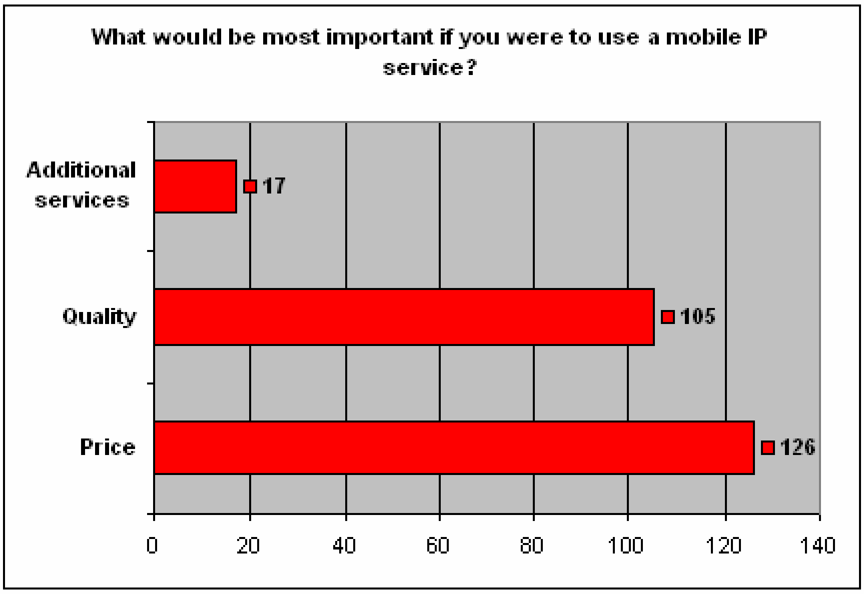
Glossary

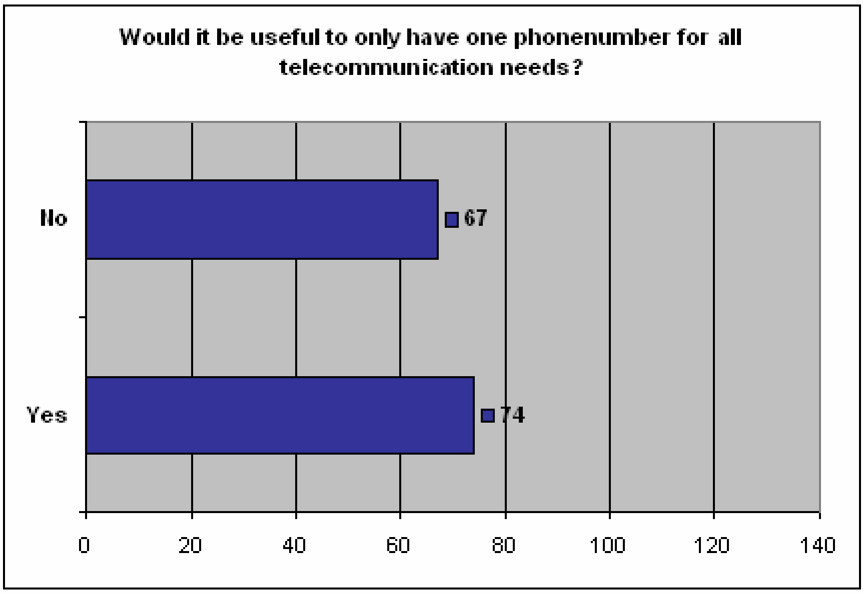
3G	Third Generation
BWCF	Breakout Control Function
CSCF	Call Session Control Function
CTM	Call Transfer Model
I-CSCF	Interrogating Call Session Control Function
S-CSCF	Serving Call Session Control Function
P-CSCF	Proxy Call Session Control Function
FMC	Fixed Mobile Convergence
HSS	Home Subscriber Service
ICM	IMS Control Model
ICS	IMS Convergence Server
ISP	Internet Service Provider
G-MSC	Gateway Mobile Switch Center
S-MSC	Serving Mobile Switch Center
GPRS	Global Packet Radio Service
GSM	Global System for Mobile communication
IM	Instant Messaging
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IP PBX	Intranet Private Branch Exchange
LMAP	LongBoard Multimedia Application Platform
MGCF	Media Gateway Control Function
MMS	Multimedia Messaging
MNO	Mobile Network Operator
MSC	Mobile Switching Center
MRF	Multimedia Resource Function
MRFC	Multimedia Resource Function Control
NeDS	Network Domain Selection
NCG	Network Convergence Gateway
OAM	Operations And Maintenance
OPEX	Operating Expenditure
PDA	Personal Digital Assistant
PSTN	Public Switched Telephone Network
PTT	Push to Talk
QoS	Quality of Service
SGSN	Serving GPRS Support Node
SIMPLE	Sip for Instant Messaging and Presence Leveraging Extensions
SIP	Session Initiation Protocol
SER	SIP Express Router
SSP	Subscriber Service Profile
SMS	Short Message Service
UMA	Unlicensed Mobile Access
UMTS	Universal Mobile Telecommunications System
VoIP	Voice over IP
WAP	Wireless Access Protocol
WLAN	Wireless Local Area Network
WI-FI	Wireless Fidelity
WiMax	World Interoperability for Microwave Access

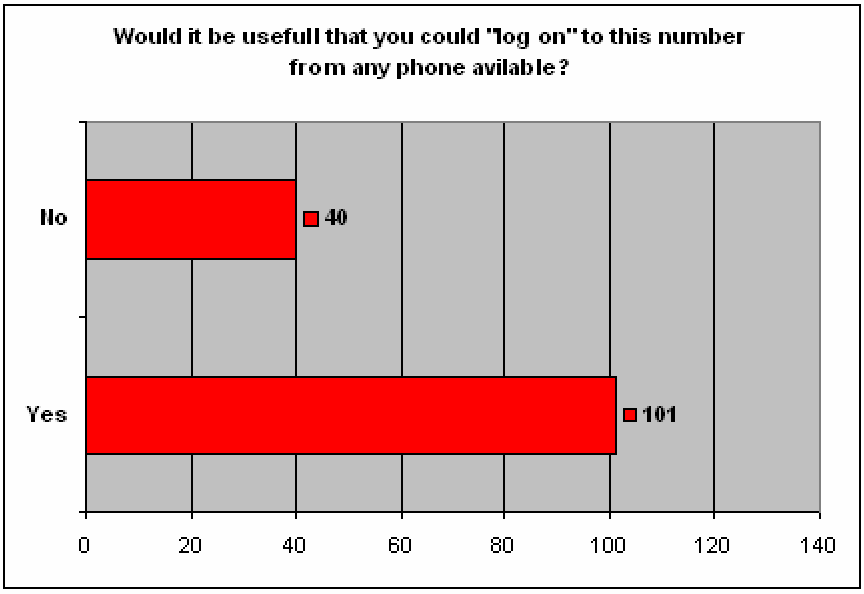
Appendix 1

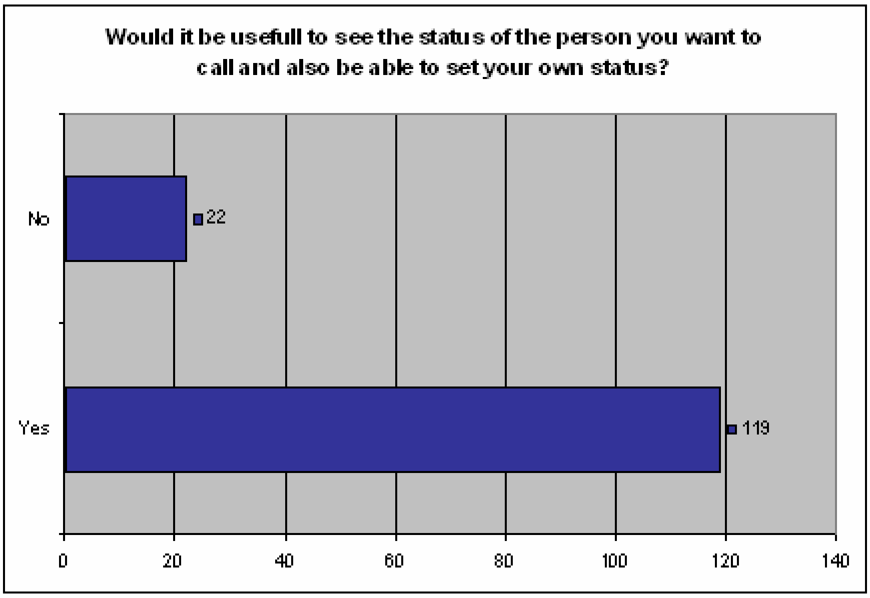

Are you answering for your self or as a business?
Business
5
Personal
136
0
20
40
60
80
100
120
140

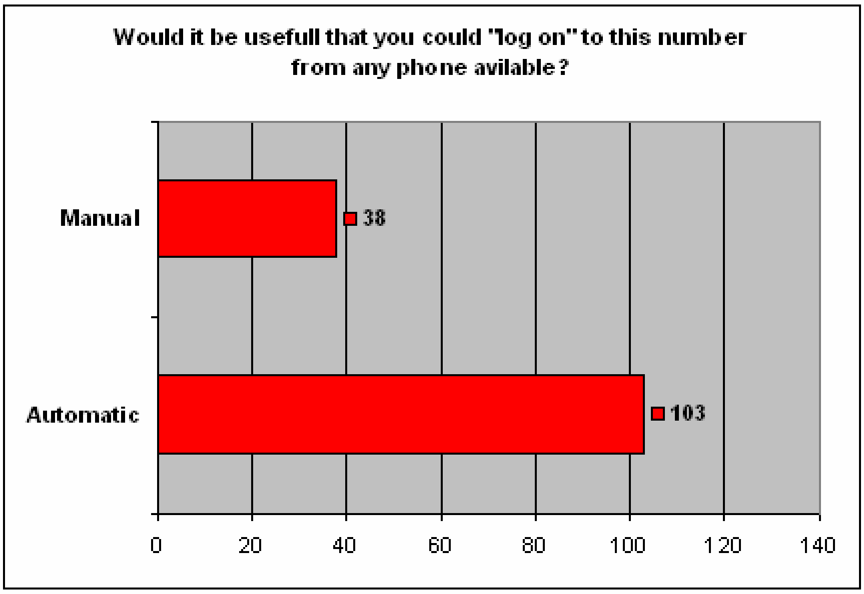

Would you be willing to install a box i your home / business
to be able to use mobile IP?
No
25
Yes
116
0
20
40
60
80
100
120
140

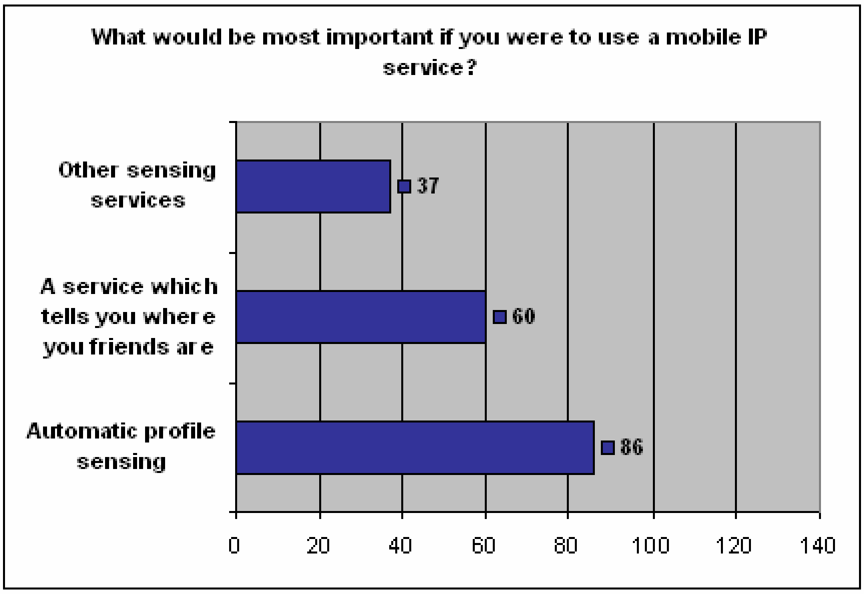

What would be most important if you were to use a mobile IP service?
Additional services
Quality
Price
17
105
126
0
20
40
60
80
100
120
140


Would it be useful to only have one phonenumber for all telecommunication needs?
No
Yes
67
74
0
20
40
60
80
100
120
140


Would it be usefull that you could "log on" to this number from any phone avilable?
No
Yes
40
101
0
20
40
60
80
100
120
140


Would it be usefull to see the status of the person you want to call and also be able to set your own status?
No
22
Yes
119
0
20
40
60
80
100
120
140


Would it be usefull that you could "log on" to this number from any phone avilable?
Manual
38
Automatic
103
0
20
40
60
80
100
120
140


What would be most important if you were to use a mobile IP service?
Other sensing services
37
A service which tells you where you friends are
60
Automatic profile sensing
86
0
20
40
60
80
100
120
140