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Migration towards Multimedia Next Generation Network

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Abstract: - The last years, the voice and the Internet access services have become a commodity, having as a result the offering from network providers, both fixed and mobile, a bunch of services. These developments are posing new requirements to existing networks, which are circuit based, and an evolution path is needed towards next generation networks, which allows the provision of multimedia services with guarantee quality of service. Mobile operators are facing the greatest challenges, since they have to reform their networks in order to provide data and multimedia services and the definition of the IP multimedia Subsystem (IMS) framework is one step towards that direction. The purpose of this paper is to shed light on significant issues, in terms of network architecture and protocols of multimedia next generation networks, allowing the definition of an evolution scenario of existing networks towards next generation multimedia networks.

Key-Words: - Next Generation Networks, Services, IP, Multimedia

1 Introduction

Streaming real-time multimedia traffic over the Internet is gaining impetus in the communications and entertainment industries. Improvements and innovations in the wide/ metropolitan, local/ access, and enterprise/ home network technologies bring cost effectively information superhighway to an increasing number of users worldwide.

At the same time, recent research efforts and developments in mobile networks like the Third Generation Partnership Project (3GPP) and 3GPP2 [1] are also fostering IP telecommunication services and multimedia services respectively, allowing mobile operators to introduce next generation communication applications and services like video telephony, live streaming etc, competing with existing Internet Service Provider (ISP) and Application Service Providers (ASPs).

The net effect of these driving forces is a set of new requirements that are placed on the evolution of the major telecommunications networks, called "Next Generation Networks" (NGN). The term NGN is very broad indeed. A formal description by ETSI's NGN Starter Group defines NGN as "…a concept for defining and deploying networks, which, due to their formal separation into different layers and planes and use of open interfaces, offers service providers and operators a platform which can evolve in a step-by-step manner to create, deploy and manage innovative services".

In general, NGN streaming applications include IP telephony, broadband data and Internet access, multimedia broadcasting and various interactive applications such as multi-party video-conference, and (near) Video on Demand etc. However, NGN has been more commonly associated with voice – a vision for the future of packet-based voice networks as part of the evolution from today's Time Division Multiplexing (TDM) circuit switched voice networks.

The purpose of this article is to investigate the different building blocks, in terms of network architecture and protocols of NGN that are capable of providing advance multimedia services with guarantee Quality of Service (QoS). Advances of different technologies, which are derived from the IP world, like the eXtensible Markup Language (XML) [2] or the Simple Object Access Protocol (SOAP) [3] influences significantly not only the provided services and applications but also the architecture of NGN, since IP becomes the convergence layer of different transport and access networking technologies. On the other hand, since NGN will be based on existing networks, different architectural issues are highlighted, providing a better insight of the migration path that should be followed towards the evolution of existing network. However, over this NGN architecture, which provides adequate bandwidth and QoS capabilities, signalling, connection establishment, capabilities exchange, and conference control functions are required. Without such capabilities the objective of providing advance multimedia services over NGN infrastructure cannot be fulfilled. Thus, special attention is given to two different alternatives, namely the H.323 and the Session Initiation Protocol (SIP) [4].

The structure of the paper is the following: section 2 presents the current and future network architectures, including some indicative evolution scenarios, whereas section 3 gives an overview of 2 different approaches, which are competing for dominance in multimedia communication, namely the H.323 protocol with its additions and extensions as well as the Session Initiation Protocol (SIP). Finally, section 4 presents the main conclusions, pinpointing issues that need further study.

2 Services and Applications of NGN

2.1 Current Networks Architecture

Current networks provide a variety of services, which enhance the traditional telephony services. However, the rate of new service creation and offering are still very slow since those network still carries the inherent weaknesses of voice networks. Existing WAN and MAN networks are based on a variety of networking technologies including ATM, SDH and WDM. However, ATM and SDH are gradually moving towards the edge of the network transforming WDM as the transport networking solution for WAN and MAN networks. The advantages of ATM like QoS support, finer bandwidth granularity, differentiating services support, robust survivability mechanisms and network management maturity are now supported by different Optical Ethernet solutions provided by many Ethernet manufacturers, thus Optical Ethernet [5] is gradually becoming the dominant solution in the metro and in the access part of networks.

Regarding wireless communications, since the commercial introduction in the early 1990s, GSM has been constantly upgraded, as evidenced by the introduction of the GPRS, Enhanced Data rates for GSM Evolution (EDGE) and Enhanced GPRS. GPRS and EGPRS allow the efficient operation of always-on data/Internet services through packet mode transmission, realizing in best case, performances of more than 100 kbit/s and 384 kbit/s respectively.

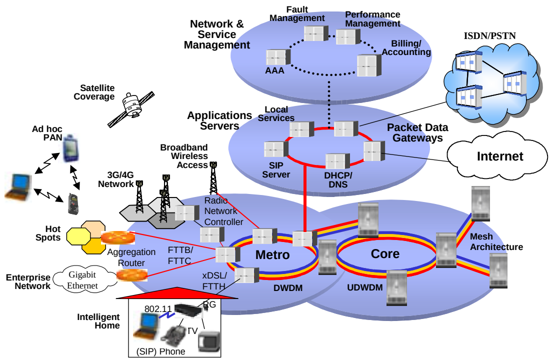

Fig. 1. NGN Architecture

2.2 Future Network Architecture

The proposed NGN architecture that could serve data, voice and video applications is shown in Fig. 1. This architecture is based on various wire-line and wireless, access, metro and core sub-networks, which are design to overcome the limitations of current voice-based networks (fixed and mobile). This is achieved by reorganizing the network architecture in order to separate the provision of the services from the network, to merge information and telephone technologies and to introduce open protocols. The main architectural difference between NGN and current ones are the convergence between the information and the telephone world, the move of intelligence from the core of the network towards the network edge and finally the lack of the need for a complex call state model, since the IP protocol is stateless.

The Wide Area Network (WAN) as well as the Metropolitan Area Network (MAN) is based on WDM networking technology, which provide huge bandwidth "pipes" in terms of wavelengths between the different Points of Present (PoPs). From a physical topology point of view both the WAN (core) and the MAN are based on ring topologies, although mesh topologies are also feasible. The fixed access part of NGN will be based on fiber access technologies (i.e Fiber to the Home – FTTH), or on copper technologies (i.e. any flavour of xDSL) or a combination of both (i.e. fiber to the building – FFTB plus VDSL inside the building in order to exploit the copper cabling of the building).

The wireless access part of NGN will be mainly based on 3G/4G mobile networks or other Fixed Broadband Wireless Access technologies like LMDS (Local Multipoint Distribution Service). It should be also noted that different variants of the WiFi technology can be used in specific physical locations (i.e hotels, airports, railway stations) complementary to 3G/4G infrastructure if bigger bandwidth requirements are needed.

It is also evident that for next generation multimedia services to work smoothly, there is the need for both vertical and horizontal roaming. The term vertical refers to access across different technologies whereas horizontal roaming is where the user may visit different WLAN ho spots owned by different operator. Furthermore, roaming capabilities between the different wireless technologies will be supported.

Common ground of all the provided service in such network architecture is the IP layer which includes routers of different size, depending on their position in the whole architecture, as well as different applications servers, which assign (IP) addresses to user's terminals, authenticate the users, store and retrieve users profile and preferences, interconnect the NGN with legacy PSTN/ISDN networks, etc.

In such environment, SIP will allow the creation, modification and termination of different sessions, like multimedia conferences, IP telephone calls, etc, providing transparency end to end. The underlying network will be used only for routing, network resources and accounting, whereas the transmission of SOAP messages over the SIP will allow one end point to communicate with another end point, making possible terminal to terminal communications. The textual nature of SIP, it will enable services to be developed rapidly in a cost effective way a variety of new services, using the XML language for their description. Finally, it should be noted that SIP will be the first step towards the migration from traditional voice network to next generation multimedia network, since it will facilitate the real integration between the data world and the voice world.

With regards to the management plane of NGN, an intelligent and advance network and service management system will allow the management of the network infrastructure in an unified way by hiding the details of each networking technology, providing a single point of view of the NGN to the network operator. Key features of such next generation Network Management System (NMS) will be the automatic end to end provisioning of the offered services with guarantee QoS based on SLA, the fast identification of network faults and the rapid recovery of the affected services in case that there are available network resource, the SLA management and the notification of the user in case of SLA violation, the performance monitoring of the provided services and the notification of the end user in case of a service degradation due to network congestion or faults, etc.

2.3 Migration Path

The deployment of NGN will not be done from scratch; on the contrary a phased migration seams to be the most viable approach. Firstly, because, the entire packet voice vision will be too radical, thus too expensive to follow, and secondly because there is little justification to replace existing voice switches and other TDM hardware that have not yet reached their end of life. However, doing nothing is not necessarily the cheapest option, as in the equipment cost one has to include i) the management, administration and maintenance cost and ii) the revenue losses and customer base reduction from the difficulties and lateness to introduce new competitive services or even lead the competition with new intelligent services. Breaking the problem into smaller pieces may indeed be the best commercial solution, as each part can proceed under its own economic constraints and timeframe. Therefore, most industry experts advocate a step-by-step approach to move towards a NGN model.

Unfortunately, the definition of a general migration path, including the “how” and “when” to migrate, is not possible. The migration strategy of every operator depends on its existing network, its plans to expand as well as its customers’ requirements. Some indicative migration steps are given below:

- Step 1 - Separation of PSTN and Data network inside the network. Such scenario permits the logical separation of the data and the voice traffic, thus there is a separate network that treats each traffic type. The points within the network that such separation can take place depend on the traffic volumes and patterns.
- Step 2 - Separation of PSTN and Data network closer to the customer: This step is similar to the previous one with the difference that the separation of the data and the voice traffic take place close to the user.
- Step 3 - Introduction of Media Gateways: The introduction of media gateways allows limited inter-working between the data and the voice world i.e. allows the translation of a telephone number to an IP address. This migration step can be treated either as a separated migration step or can be also combined with any of the previous 2 steps.
- Step 4 - Evolution of Class 5 switch to a Class 5 soft-switch: Class 5 soft-switch can serve different type of terminals (standard phones, IP phones, PCs, etc) connected either directly to the data network or through media gateways. This step allows the evolution from step 3 towards a real NGN.

3 Signaling Protocols: H.323 or SIP?

3.1 H.323

Over this NGN architecture, which provides adequate bandwidth and QoS capabilities, signalling, connection establishment, capabilities exchange, and conference control functions are required. Currently two standards are competing for dominance: the H.323 protocol suite by ITU-T, and the Session Initiation Protocol (SIP) by IETF. Fig. 2 shows the potential session signalling & control protocol stack. The H.323 started out as a protocol for multimedia communication over a LAN segment without QoS guarantees, but has evolved in order to fulfil the more complex needs of Internet telephony. The ITU H.323 series of recommendations includes H.245 for control, H.225.0 for connection establishment, H.332 for large conferences, H.450.1 H.450.2 and H.450.3 for supplementary services, H.235 for security, audio/video codecs (e.g. G.711 for audio and H.261 for video), and H.246 for interoperability with circuit-switched services. H.323 is based on precedent ITU multimedia protocols, including H.320 for ISDN, H.321 for BISDN, and H.324 for GSTN terminals. The encoding mechanisms, protocol fields, and basic operation are somewhat simplified versions of the Q.931 ISDN signalling protocol.

The H.323 call signalling procedure begins when an originating H.323 endpoint issues an admission request (ARQ) to the H.323 Gate-Keeper (GK) in its zone. The GK provides services, such as address translation, Remote Access Server (RAS) control, call redirection and resource management. After the endpoint receives a confirmation message (ACF) from the gatekeeper, the call setup procedure continues with a SETUP and CONNECTION message exchange. Finally, both endpoints follow the H.245 capability exchange procedure: they exchange terminalCapabilitySet messages and open media channels. Clients can reduce the signalling overhead by using the Fast Connection procedure, which allows them to start media communication after one round-trip message exchange instead of three. The Fast Connection procedure is initiated by including a fastStart element in the SETUP message.

The FastStart element carries the proposed media channel description, OpenLogicalChannel, which identifies the media capability of the originating endpoint.

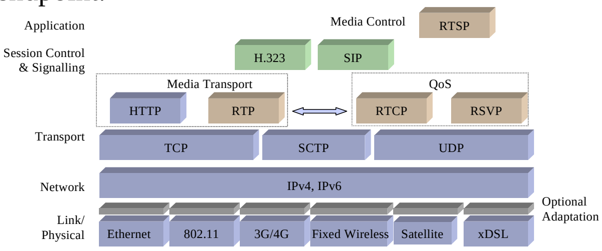

Fig. 2. Session Signaling & Control Stack

(RTSP: Real-Time Streaming Protocol, SIP: Session Initiation Protocol, RTCP: Real-Time Control Protocol, RTP: Real time Transport Protocol, RSVP: Resource Reservation Set-up Protocol, HTTP: Hyper Text Transfer Protocol, TCP: Transmission Control Protocol, SCTP: Stream Control Transport Protocol, UDP: User Datagram Protocol)

3.2 SIP

The Session Initiation Protocol (SIP) takes a different approach by reusing many of the header fields, encoding rules, error codes, and authentication mechanisms of HTTP. In both cases, multimedia data will likely be exchanged via RTP (Real time Transport Protocol), so that the choice of protocol suite does not influence Internet telephony QoS.

A generic SIP operation involves a SIP User Agent Client (UAC) issuing an invitation, a SIP proxy server acting as end-user location discovery agent, and a SIP User Agent Server (UAS) accepting the call. A successful SIP invitation consists of two messages: an INVITE followed by an ACK. The INVITE message contains a session description that informs the called party what type of media the caller can accept and where it wishes the media data to be sent. The SIP messages use URLs addresses and a human-readable, text-based encoding message format, based on the HTTP. The Session Announcement Protocol (SAP) and the Session Description Protocol (SDP) support the establishment of multiparty conferencing sessions. SAP defines the procedures for advertising conferencing sessions by periodically multicasting information about active sessions. SDP supports the description of multimedia sessions, including the specification of preferred media types and scheduling information.

SIP has a clear preceded over H.323 as it has been selected by the 3GPP and 3GPP2 as the basis for their IP based Multimedia Subsystem. However, for NGN packet-based voice networks, there will be a need for interoperability between SIP and H.323 signaling [6] for interconnection between different existing VoIP networks and PSTN/ISDN legacy networks.

Table 1 illustrate an overview of the 2 protocols in terms of complexity, extensibility, scalability, and services support aspects.

Table 1: Comparison Overview

Criteria	**H.323**	**SIP**
Complexity	Complex	Reduced complexity
Message Set	Binary ASN.1 encoding.	Text format, llogically numbered/structured responses
Message parsing/ Debugging	Specialized tools	Simple Tools. Reusability of HTTP code.
Methods for implementing services	Umbrella of ITU-T protocols	Re-use Internet protocols
Extensibility	Extensible	More Extensible
Backwards Compatibility and features evolution	Supported. Suffers from functionality redundancy.	More flexibility
3rd party services	Less Ability – Complex ASN.1	Higher Ability – text formats and extension headers.
Modularity	Umbrella protocol standard.	Modular design
Scalability	Installed base designed for reliable transport	Designed for it
Wide Area Support	Yes. May handle large number of calls	Better
Message	Smaller binary messages	Text formatted, larger messages
CPU processing	More processing	Less processing
QOS	Supported	Supported
Services	High	Low
Supported Services	H.323 more explicitly defined (standardized)	SIP defined in white papers/Internet drafts/RFC
Capabilities Exchange	Better for media – worse for signalling extensibility	Worse for media – better for signalling extensibility
Personal Mobility	Location based services still ongoing	Designed with mobility. Location based services still ongoing
Legacy interoperability	H.246	Draft status H.323
Security	H.235 added later. Worse for firewall traversal.	Designed with security. Better for firewall traversal.

4 Conclusions

Multimedia NGN is the next big revolution in telecommunications. Starting from the description of current networks architecture, this article provides an insight of future networks architecture. The major concern for network operators of how to migrate from their existing networks to the new ones, in which the IP protocol will be inevitably the unification layer, while minimizing their costs of transition and maximizing the benefits of those new networks, is also covered by presenting an indicative step-by-step migration path. However, the study of NGN architectural aspects is only one side of the whole picture. The investigation of next generation service and applications are also important, therefore this article also discusses the most representative enabling technologies for the definition of such services and applications, their interactions, the development of them and finally the control of them. The latter is investigated in details by comparing two different approaches namely the H.323 and SIP protocols.

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