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Contents

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Services and Protocols for Advanced Networks (SPAN).

Introduction

This work has been carried out by STF 234 under the control of the European Commission. The objective of the presented Technical Report is to evaluate the requirements for standardization, identify existing standards and to identify the standardization gaps with respect to the technology for Broadband Multimedia Services.

The convergence of telecommunications, radio communications the information technology and home electronics, and also the convergence of broadcasting and interactive applications, which is coupled with the variety of networks, including the new ones like IP based networks and mobile networks, brings the roll out of new broadband multimedia services.

With the emergence of high-speed networks, the European society will request real-time multimedia communications as an extension of existing one-media systems. This apply for an all range of applications and services in key areas of the eEurope 2005 initiative, such as e-government, e-learning, e-health and e-business. Diversity of emerging new broadband multimedia services require to speed up the standardization process in the whole area of multimedia-related and broadband technologies in such a way the standards fully respond to user requirements in terms of easy to use, accessibility, mobility, security, quality of service, end-to-end interoperability, etc.

Broadband multimedia applications and services need to be independent of the networks they operate across. The impetus for this has been the growth in the use of the Internet with its concept of "available anywhere at anytime". An open network architecture has also to ensure that new broadband multimedia applications can be readily configured to meet user needs using available network resources in a network-independent way. In other words, the architecture for the support of broadband multimedia services has to ensure that users can get the information content they want, in any media, anytime, anywhere, over any facilities. This context has generated a need for a new networks model called NGN (Next Generation Networks).

Considering the facts mentioned above, the following approach has been taken for the present document:

Clause 4 introduces broadband multimedia services and applications with respect to the definition, classification and identification of possible services. It also provides the multimedia services reference model.

Clause 5 identifies technical requirements for the support and deployment of broadband multimedia services. As the basis for doing this, the NGN model with its layering architecture has been chosen. The technical requirements are thus identified with respect to the NGN access, transport, control and service layer. However, there are also technical issues that are affecting all NGN layers. They are specified in clause 5.6 and include addressing, naming and numbering, media coding, quality of service, security, network management, interoperability and multimedia mobility.

In clause 6, an overview of running activities within different standardization bodies related to broadband multimedia services and technologies and to NGN is given.

Clause 7 identifies standardization gaps with respect to the technology for broadband multimedia services and NGN. The standardization gaps are classified with respect to the structure of the present document as mentioned above.

1 Scope

The present document analyses technical requirements for the support and deployment of Broadband Multimedia Services in Europe. It evaluates the requirements for standardization, identifies existing standards and standardization gaps with respect to the technology for Broadband Multimedia Services.

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3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AAA	Authentication, Authorization and Accounting
AALn	ATM Adaptation Layer n
ADM	Adaptive Delta Modulation
ADPCM	Adaptive DPCM
ADSL	Asymmetric Digital Subscriber Line
AH	Authentication Header

ANMP	Ad-hoc Network Management Protocol
AP	Access Point
API	Application Programming Interface
ASAP	Application Specific Access Profile
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
BA	Basic Access
BICC	Bearer Independent Call Control
BS	Base Station
B2C	Business to Customer
CA	Certification Authority
CAP	Carrierless Amplitude/Phase modulation
CAT	Common Authentication Technology
CATV	CAble TeleVision
CCXML	call Control XML
CDMA	Code Division Multiple Access
CIDR	Classless Inter-Domain Routing
CMIP	Common Management Information Protocol
CSn	Capability Set n
CSN	Circuit Switched Network
DAVIC	Digital-Audio VIsual Council
DHCP	Dynamic Host Configuration Protocol
DMAP	DECT Multimedia Access Profile
DNS	Domain Name System
DOCSIS	Data Over Cable Service Interface Specification
DPCM	Differential Pulse Code Modulation
DPRS	DECT Packet Radio Service
DSL	Digital Subscriber Line
DVB	Digital Video Broadcast
EDGE	Enhanced Data for GSM Evolution
ESP	Encapsulating Security Payload
FDD	Frequency Division Duplex
FDDI	Fibre Distributed Data Interface
FEC	Forwarding Equivalence Class
FR	Frame Relay
FTP	File Transfer Protocol
FTTP	Fibre-To-The-Building
GMM	Global Multimedia Mobility
GPRS	General Packet Radio Service
GSS	Generic Security Service
HA	Home Agent
HAPS	High Altitude Platform Station
HDTV	High-definition Digital TeleVision
HLR	Home Location Register
HTTP	Hyper Text Transport Protocol
IP	Internet Protocol
ISUP	ISDN User Part
IUA	ISDN User Agent
LAN	Local Area Network
LEO	Low Earth Orbit
LMDS	Land Microwave Distribution System
LMMP	LAN MAN Management Protocol
M2PA	Message transfer part level 2 Peer-to-peer Adaptation layer
M2UA	Message transfer part level 2 User Adaptation layer
M3UA	Message transfer part level 3 User Adaptation layer
MAC	Media Access Control
MAN	Metropolitan Area Network
MCU	Multipoint Control Unit
MG	Media Gateway
MGC	Media Gateway Control
MGCP	Media Gateway Control Protocol
MHP	Multimedia Home Platform

MIME	Multi purpose Internet Mail Extension
MMDS	Multichannel Multipoint Distribution System
MPEG	Moving Picture Expert Group
MPLS	Multi-Protocol Labelling System
MR	Mobile Router
MTP	Message Transfer part
MVDS	Microwave Video Distribution System
MWS	Multimedia Wireless System
NAPTR	Naming Authority PoinTer Resource record
NGN	Next Generation Network
OLC	Open Logical Channel
OS	Operating System
OSA	Open Service Architecture
PAM	Pulse Amplitude Modulation
PKI	Public Key Infrastructure
PON	Passive Optical Network
POTS	Plain Old Telephone Service
PSTN	Public Switched Telecommunication Network
QoS	Quality of Service
RADSL	Rate-Adaptive Digital Subscriber Line
RAS	Registration Admission Status
RSVP	ReSource reserVation Protocol
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SCE	Service Creation Environment
SCF	Service Capability Feature
SCN	Switched Communication Network
SCTP	Stream Control Transmission Protocol
SDH	Synchronous Digital Hierarchy
SDSL	Symmetric Digital Subscriber Line
SHDSL	Symmetric High Digital Subscriber Line
SDTV	Standard definition Digital TeleVision
SIP	Session Initiation Protocol
SIP-T	Session Initiation Protocol-Telephony
SG	Signalling Gateway
SMIME	Secure MIME
SMS	Short Message Service
SMTP	Simple mail Transfer Protocol
SNMP	Simple Network Management Protocol
SOAP	Simple Object Access Protocol
SOHO	Small Office Home Office
SS7	Signalling System No.7
SSL	Secure Socket Layer
STP	Shielded Twisted Pair
SUA	SCCP User Adaptation layer
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TLS	Transport Layer Security
TSAP	Transport layer Service Access Point
TIF	Text Image Format
UCI	Universal Communication Identification
UDDI	Universal Description, Discovery and Integration
UDP	User Datagram Protocol
UHF	Ultra High Frequency
UM	Unified Messaging
UMTS	Universal Mobile Telecommunications System
URL	Uniform Resource Locator
UTP	Unshielded Twisted Pair
UTRA	Universal Terrestrial Radio Access
VC	Virtual Channel
VCI	Virtual Channel Identifier
VDSL	Very high-speed bit rate Digital Subscriber Line

VHE	Virtual Home Environment
VHF	Very High Frequency
VP	Virtual Path
VPI	Virtual Path Identifier
VPN	Virtual Private Network
VRML	Virtual Reality Modelling Language
WAP	Wireless Application Protocol
WDM	Wavelength Division Multiplexing
WLAN	Wireless LAN
WSDL	Web Service Definition Language
WTLS	Wireless TLS
XML	eXtensible Markup Language

4 Description of broadband multimedia services

4.1 Definition of broadband multimedia services

4.1.1 Definition of broadband

The term "broadband" has often a different meaning. Various definers of broadband have assigned a minimum data rate to the term.

- Newton's Telecom Dictionary: "...greater than a voice grade line of 3 KHz...some say (it should be at least) 20 KHz".
- Jupiter Communications: at least 256 Kbps.
- IBM Dictionary of Computing: A broadband channel is "6 MHz wide".

There are many criteria to define "broadband". Among them, the primary ones are ("A Broadband Wireless Framework for 2003" - see bibliography):

- Maintaining an Always-on-Connection. Users should be able to power up their computers or simply launch their browsers and have a connection for as long as they are on-line.
- Bandwidth Speed must include a minimum of 384 kbit/s downstream and 128 kbit/s upstream to be considered broadband today, relative to available cable modem, DSL, and satellite offerings, but also must be able to extend upwards in the future.

When defining the term "broadband", the several aspects have to be taken into account:

- digital content: thanks to the matured digital technologies all the types of information (voice, text, audio, video) can be converted and transmitted in a digital form;
- packet-based: IP flow transport in native IP, or on ATM, with a progressive convergence to IP;
- data rate: "…the capability of supporting, in both the provider-to-consumer (downstream) and the consumer-to-provider (upstream) directions, a speed (in technical terms, "bandwidth") in excess of 200 kbit/s in the last mile…" (FCC report);
- interactivity: the possibility of a dialog between the end-users (regardless of whether they are humans or machines).

4.1.2 Multimedia service

Multimedia services are telecommunication services that handle two or more types of media in a synchronized way from the user's point of view. A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources and users within a single communication session [1].

4.1.3 Multimedia application

A multimedia application is an application that requests the handling of two or more representation media (information types) simultaneously which constitute a common information space. Examples are cooperative document editing, long distance meetings, remote surveillance, medical document remote analysis and teletraining [1].

4.1.4 Broadband multimedia service

"Broadband" multimedia is sometimes called "streaming" multimedia because the services, or the "content" that is delivered via broadband networks is digitized, and received by users of the content in continuous real-time "streams". Broadband multimedia services will put the user in total control by enabling personal, custom, on-demand viewing of any type of content, like infotainment, entertainment, e-learning, video games, etc. Furthermore, broadband multimedia services will allow individual users to easily create their own content, personalize it, and distribute it for viewing on TVs, PCs, remote laptops, mobile phones and other wireless devices around the world, instantly. They will also create new revenue streams for operators, media companies, and service providers through enhanced usage of existing networks.

Broadband multimedia services is the seamless, customized "on demand" creation and delivery of multimedia services to homes, businesses, and mobile users, including entertainment services (movies, interactive games, broadcast TV), infotainment (e-learning, online training) through high-speed broadband access and high-speed core packet networks (IEC on-line tutorials: "Broadband Media Services"- see bibliography).

4.2 Classification of broadband multimedia services and applications

4.2.1 Criteria for classification

Based on the definition given in [1], multimedia services are telecommunication services that handle two or more types of media in a synchronized way from the user's point of view. Nowadays, several types of media are conveyed by networks:

- text (ASCII, etc.);
- audio;
- graphics (2D, 3D);
- images (still or animated);
- video;
- data.

These traffics have requirements inherent to the nature of the services that issue them.

There are several criteria that might be taken into account to classify the broadband multimedia services.

- A user point of view and a network provider point of view [1]:
  - From the end user's point of view, a multimedia telecommunication service is the combination of telecommunication capabilities required to support a particular multimedia application. Such a service is usually considered to be independent of the network(s) providing these capabilities.
  - From the network provider's point of view, a multimedia telecommunication service is a combination or set of combinations of two or more media components (e.g. audio, video, graphics, etc.) within a particular network environment in such a way as to produce a new telecommunication service. This telecommunication service is considered to be fully dependent on the specific capabilities of the networks utilized.

- Different types of users (1):
  - Residential.
  - SOHO (Small Office Home Office).
  - Corporate.
- Different types of users (2):
  - PC users.
  - TV users.
- Character of a service:
  - based on ITU-T classification of the B-ISDN services [2] (does not cover only multimedia services):
    - **conversational services** providing the means for bi-directional dialogue in real-time (e.g. videotelephony, (personal communication), videoconference (group communication), videosurveillance (monitoring elderly people), video/audio information transmission services (video/audio dialogue), multiple sound programme signals (multilingual commentary channels), high-speed unrestricted digital information transmission service (computer interconnection), high volume file transfer services (data file transfer), high-speed teleaction (real-time control/alarms), high-speed telefax, high resolution image communication services (remote games and game networks, medical images, etc.));
    - **messaging services** providing the means for bi-directional dialogue using store-and-forward mechanisms (e.g. video mail service, document mail service, etc.);
    - **retrieval information** stored in information centres, where the user accesses the information on demand (e.g. video retrieval service (education), high resolution image retrieval service (remote publishing), document retrieval service (advertising), data retrieval service (telesoftware));
    - **distribution services** providing a continuous flow of information, which the user can view from time to time, without individual presentation control (the most obvious example being TV broadcasting, other examples are pay-TV, document distribution services (electronic newspaper), high-speed unrestricted digital information distribution services, video distribution services with user individual presentation control, full channel broadcast videography (remote education and training/news retrieval), home control, personal services (monitoring of health, local teleshopping, banking services, etc.).
  - based on the UMTS QoS classes:
    - **Conversational** (e.g. conversational voice, videophone, on-line games).
    - **Interactive** (e.g. database retrieval, Web browsing, server access).
    - **Streaming** (e.g. streaming audio, streaming video).
    - **Background** (e.g. e-mail, SMS, download of databases, reception of measurement records).

4.2.2 Multimedia services reference model

Multimedia services reference model can be organized in four levels [1]:

- **Application level:** The level at which the essential functional characteristics of an end-user application are described from the end-user's point of view irrespective of the underlying technical aspects of the services or particular network solution.
- **Service level:** The level at which the basic set of communication services or support tools required to satisfy the functional requirements of the application level are identified. The overall service principles (for performance, QoS, security, charging, intercommunication) are defined and described. A service is built up by combining communication tasks and organizing their interaction. The service level may contain a description of how to find where end-user and terminals are located.

- **Communication task level:** The level at which the specific communication tasks required to build the services are defined. A communication level task is a functional entity of a multimedia service which performs its communication features. It handles a set of media components in a synchronized way, in order to convey and control complex information types.

- **Media component level:** The level at which the multimedia aspects of the services are made apparent through the identification and description of the individual (monomedia) components related to a single information type, such as audio, video, etc.

In addition to the levels which are related to the communication plane, control and processing functions are required for operating the service (see table 1). In other words, the service platform also uses elements of control functions (Middleware service elements in the ITU-T Recommendation F.700 terminology) that control the various levels of the communication plane or perform appropriate processing of the transmitted information. The elements of control functions may thus interact with a specific level (service, communication task, media component), or they may interact with two or more levels (e.g. for intercommunication, either at the lowest level between terminals to allow a user to make different types of calls from the same terminal, or at a higher level to allow the user to combine the functions offered by different type of services and thus increase the range of possible applications).

Table 1: Functional model of the service platform (source: ITU-T Recommendation F.700 [1] figure 2)

Service platform	
Communication plane	**Control and processing plane**
services	middleware service components
communication tasks	
media components	

The functions provided by the specific levels:

- **at the service level:** the general functions related to the call and to the type of service (call establishment, security, charging, etc.);

- **at the communication task level:** the functions related to the configuration, the time aspects, the linking of media components (e.g. transfer, storage, multipoint access, switching or multiplexing signals);

- **at the media component level:** the functions related to each independent medium (e.g. capture, coding, presentation, etc.).

4.2.2.1 Communication tasks

A multimedia service can be decomposed into a set of communication tasks, each of them being, separately or not, manipulated by the user and/or service provider. From a reverse point of view, a communication task can be viewed as a means of bringing together the media components that are related each to other for the purpose of the service.

The communication tasks are described using attributes and their possible values. So far, the following attributes and values have been identified [1].

Table 2: Communication task attributes (based on ITU-T Recommendation F.700 [1] table 1)

Attributes	Possible values
Communication configuration	Point-to-point/point-to-multipoint, multipoint-to-point, multipoint-to-multipoint
Symmetry of information flow	Unidirectional/bidirectional-symmetric/bidirectional-asymmetric
Transmission control entity	Source/sink/source and sink/third party
Communication delay	Real-time *(e.g. conversational services, interactive video games)* Near-real time *(e.g. retrieval services)* Non-real time *(e.g. storage services)* Specified time *(e.g. VoD, retrieval services with a response within a defined time)*
Mandatory media components	Audio/video/text/still picture/graphics/data
Optional media components	Audio/video/text/still picture/graphics/data/none
Media component interrelations	Synchronization between: audio and video (lip synchronism, location related synchronism) audio and text (voice synthesis) text and video/still picture/graphics (subtitles synchronized with images) graphics and video Symmetry between media components of the same time to allow for bidirectionality Conversion between information types (or media components)
Time continuity	Isochronous: Isochronous transmission is necessary if the user terminal equipment has no buffering capabilities, or if the capturing device does not tolerate interruptions or variations in transmission speed or does not have a large enough storage capacity. Non-isochronous

Based on the combination of the attributes "communication configuration", "symmetry of information flow", and "transmission control entity", the six basic communication tasks have been identified [1].

Table 3: List of communication tasks (based on ITU-T Recommendation F.700 [1] table 3)

Communication task	Communication configuration	Symmetry of information flow	Transmission control entity
Conversing	Point-to-point	Bidirectional	Source and sink controlled
Sending	Point-to-point	Unidirectional	Source controlled
Receiving	Point-to-point	Unidirectional	Sink controlled
Distributing	Point-to-multipoint	Unidirectional	Source controlled
Collecting	Multipoint-to-point	Unidirectional	Sink controlled
Conferencing	Multipoint-to-multipoint	Bidirectional	Source and sink controlled

The list does not take into account the attributes such as time aspects, third party control, etc. which may enhance the list in further communication tasks.

4.2.2.2 Media components

Media components are those parts of communication service which provide the communication capabilities related to a single information type. They provide the necessary functions for user information handling, such as information capture, presentation, storage, transfer and post-processing. The following are examples of media components suited for presentation to human beings:

- **Audio:** In the media component audio, a source produces an audio signal which than may be coded, compressed, stored and transmitted. At the other end, the signal is decoded and presented to a human user.

- **Video:** In the media component video, a video source produces a video signal carrying a moving picture, which can be then coded, compressed, stored and transmitted. At the other hand, the signal is decoded and presented on a screen.

- **Text:** Text is a representation medium consisting of formatted characters. It is stored and transmitted as a sequence of codes. Although it may be displayed on the same screen as video and still pictures, it requires decoding into specific fonts for presentation to the user, whether on the screen or on a paper. The input is through a keyboard. The output may be a printer or a screen.

- **Still picture:** Still picture is a pixel-based representation medium. They are initially digitized as bit maps, i.e. as one or several bits allocated to each pixel and coding for its colour. Because of the large volume of data generated, the data is then usually compressed by using the correlation between different parts of the picture and often by accepting a limited amount of degradation through loss of information. Depending upon the application, the same type of transducers as for video may be used (cameras, screens), or scanners and printers.

- **Graphics:** Graphics is a representation medium consisting of geometrical objects featured by their positions, shapes and colours. It is stored and transmitted as a set of codes and parameters. Although it may be displayed on the same screen as video and still pictures, it requires decoding into specific geometrical figures for presentation to the user, whether on a screen, or on a paper. The input may be through a graphics tablet, an electronic pencil, some other two-dimensional transducer, or a dedicated graphic software on a PC or workstation. The output device may be a printer or a screen.

- **Data:** Data is a medium (see note) that consists of a sequence of bits often organized in files, i.e. finite sequences. Two types of files are defined upon their contents: software files (use to store or to download software), data files (finite sequences of bits arranged to a set of rules and associated with a given software, which is needed to generate it, modify it or use it).

NOTE: Data files are not a media component according to the ITU-T Recommendation F.700 [1] definition, because they are not associated with any unique perception medium. But, they may support various types of media components for storage and transmission, they may also contain data for various applications.

4.2.2.3 Control and processing functions

Control and processing functions are associated with the service and communication tasks level, possibly but not necessarily with the media component level (for example, the coding and decoding process belongs to the media component level).

The control features are usually embodied in protocols independent from transmission systems. Their functional descriptions are independent from the basic protocol, but the set of specific commands and messages associated with the protocol has to support the required functions.

The examples of control and processing functions include:

- security (authentication/non-repudiation, privacy, integrity);
- directory;
- reservation;
- call control (call set-up, call modification, quality negotiation);
- charging and billing;
- media selection;
- conference control (conference management, multipoint control);
- searching (browsing, navigation);
- mailing (sending mail, retrieving mail);
- polling;
- application control (navigation in a document, device control);
- processing (selection, assembly, translation, media conversion);
- storage;
- replication;
- intercommunication.

4.2.2.4 Network aspects

It should be noted that the ITU-T Recommendation F.700 [1] Multimedia service reference model and the following series of General Service Recommendations for generic services (e.g. for Multimedia Conferencing Services [3], Multimedia Conversational Services [4], etc.) provide functional descriptions of services that are independent from the networks that support them.

However, different applications have various constraints not only with respect to the network parameters. For example, with respect to the network delay and the network loss, applications can be roughly classified into four categories:

- Applications that are delay-sensitive and loss-sensitive (e.g. interactive video).
- Applications that are delay-sensitive but tolerant of moderate losses (e.g. voice).
- Applications that are sensitive to loss of data but tolerant to moderate delays (e.g. interactive data).
- Applications that are relatively tolerant to both delay and some limited loss of information (e.g. file transfer).

Moreover, different types of continuous media will have different level of constraints on:

- Bandwidth.
- Jitter (that may be depend on the network jitter, jitter buffer, gateway buffer, depending on the network configuration).
- Guarantees that levels of service can be maintained.

For example, video communications require high throughput guarantees but telephone audio requires only modest bandwidth. Uncompressed video is highly tolerant of communication errors whereas compressed voice cannot tolerate high error rates; file transfer should be 100 % error free.

Distributed real-time and multimedia applications require a communication service to transfer time-constrained audio samples, video frames, sensor inputs, or actuator commands over the network. Time constraints are necessary for audio and video streams when human beings are involved. Human beings are very sensitive to delay and jitter but also able to tolerate transient delay and jitter violations.

With respect to interactive communications, we can distinguish:

- **live communications**, where producer and consumer are human beings;
- **playback communications**, where the producer is a playback device at which audio/video data is stored and only the consumer is a human being.

Network connections might be:

- **unicast:** a connection is uni-cast when it is established between two users (it is a point to point connection);
- **broadcast:** a connection is broadcast, when all the connected users may be reached by the same message;
- **multicast:** a connection is multicast if it enables a particular group of users to communicate;
- **any-cast:** a connection is any-cast if a message may be received by one (any) user belonging to a group.

Some applications (such as file transfer) require only uni-cast connections, but others (such as video conferencing) generally require broadcast or multicast connections.

4.2.3 Identification of generic service capabilities

This clause provides an overview of existing and possible future generic services and applications. It has to be noted that the services in the overview given below have been identified from a user's point of view. It also shows positioning of broadband multimedia services within the overall service landscape. This clause does not intend to identify all services but rather to identify generic service mechanisms over which various services and applications can be built and instantiated (therefore the list of services does not include traditional PSTN/ISDN-based supplementary services).

1) User services

- Community communication tools
  - **Directory:** On-line directory services, similar to "yellow" pages.
  - **Co-browsing:** The possibility to push and share a screen or a content between end-users.
  - **Remote control:** The capability to control or monitor remotely end-users' terminal equipment and/or applications.
  - **Document sharing:** The capability to share presentations and documents among the end-users.
- User's environment services
  - **Personal data services:** The ability of the end-users to manage their personal profiles, calendar, contact list, etc.
    - User profile services.
    - Diary services.
    - Address book.
    - Communication services availability.
- **Context management services:** The ability for the end-users to maintain, operate and manage services
  - **Presence services:** The capability to support management of presence information between watchers and present users, in order to enable applications and services to make use of presence information. In the context of mobile services, presence and availability technologies provide the ability to determine the event in which a mobile user is present in a certain location and/or available for certain events to take place such as mobile messaging, games, and other location based services (ETSI User Group; User Interoperability Criteria - see bibliography).
  - **Instant messaging services:** The capability to maintain a list of people the end-user wishes to interact with. He/she can send messages to any of the people in the list, often called a buddy list or contact list, as long as that person is online. Sending a message opens a small window where the initiating end-user and his/her correspondent can type in messages that both of them can see (ETSI User group; User Interoperability Criteria - see bibliography).
  - **Location-based services:** The capability to provide customized service provision depending on the customer positioning. Such positioning may be either GPS based or network based. The network based positioning typically rely on various means of triangulation of the signal from cell sites serving a mobile phone. The four major categories of location based services include location based information, location sensitive billing, emergency services and tracking (ETSI User group; User Interoperability Criteria - see bibliography).
  - **Notification services:** The possibility for registered end-users to obtain events (autonomous messages), e.g. alarm messages, performance reports, events occurring in a particular system, etc.
  - **Dynamical management of the session content:** The possibility to search for, sort, and filter the session content.

2) Services for inter-personal communication

- Non-real-time communication services
  - **Monomedia messaging:** The capability to send monomedia messages (voice, or text, or video), to receive messages, to request confirmation of receipt, to modify the content as well as the recipient of the message, to reject an outgoing and/or incoming message and to re-route a message.

- **Unified Multimedia messaging:** The capability to bring together all messaging media such as voice messaging, SMS, and other mobile text messaging, e-mail and facsimile into a combined communication experience. Minimally, the communication experience will take the form of a unified mailbox and/or alert service, allowing the end-user to have a single source for message delivery, repository, access and notification (ETSI User Group; User Interoperability Criteria - see bibliography).

- Real time communication services
  - Establishment of connectivity
    - Audio.
    - Video.
    - Text.
  - Point-to-point
    - Voice.
    - Videotelephony.
    - Text chat.
  - Multipoint
    - Audioconference.
    - Videoconference.
    - Text chat in a conference.
    - Edutainment and Infotainment (e.g. gaming, travel information services, educational services, B2C services).
    - Collaborative working.
    - E-commerce, m-commerce services.

3) Content delivery services

- **Streaming media (broadband media) services (see note):** The capability to receive services or "content" that is delivered via broadband networks in continuous real-time "streams" (e.g. video streaming, Web broadcast).
- **Content adaptation services:** The capability to present the content adaptively with regard to the delivery platform (end-user device), the user (access type and service profile and preferences) and usage context (time and position, etc.).
- **File downloading (e.g. ftp).**
- **Network storage and processing services** (e.g. provision and management of information storage units, file servers, terminal servers, OS platforms, etc).
- **Push services** ("to plug-in and get a content"): The capability to receive "the content" without a necessity to explicitly request for it. The "content" sent to the end-user is based on a user profile and is sent to him asynchronously, as soon as it becomes available.
- **Peer-to-peer services** (e.g. sharing of computing resources and files).

4) Broadcast services

Broadcast services are those vision, multimedia and data services intended to be for use by the public, including of those using access control or interactivity. They typically use an asymmetrical distribution infrastructure that allows high capacity information downloading to the public with a low capacity backward link to the service providers (e.g. as in UMTS Report No.11 - see bibliography). The service examples are contribution and delivery of TV/radio and data, audio broadcasting, Video on Demand, interactive broadcast services, etc.

NOTE: It might happen that in the context of converging media technologies, streamed audio and video content obtainable via IP-based networks will be regarded as a broadcasting service.

5 Technical requirements for the support and deployment of broadband multimedia services

5.1 Architecture

Broadband multimedia applications and services need to be independent of the networks they operate across. The impetus for this has been the growth in the use of the Internet with its concept of "***available anywhere and anytime***".

Additionally, an open network architecture has to ensure that new broadband multimedia applications can be readily configured to meet user needs using available network resources in a network-independent way. In order words, the architecture for the support of broadband multimedia services has to ensure that users can get the information content they want, in any media, anytime, anywhere, over any facilities.

This context has generated a need for a new networks model called **NGN (Next Generation Networks)**. The NGN concept is based on a progressive evolution to end-to-end "***all IP***". It has been designed in such a way to be able to provide all the types of emerging multimedia services and to keep at the same time existing services provided by current CSN and IP-based networks. The NGN is essentially characterized by the separation of the transport and control layer and the fact that the former is based on packet technology. The aim is to build a converged network where all media types share the same transport infrastructure. The other aim of NGN is that it opens the way for new breed of services.

The principal characteristics of the NGN can be defined as follows:

- A unique and shared network for all types of accesses and services.
- Packet-based transfer transparent to users.
- A core network architecture divided into three layers: Transport, Control and Services. This brings also the necessity of the separation of control functions for bearer control, call/session control and service/application control functions and leads to the open distributed control architecture that replaces the classical "monolithic" switch.
- Decoupling of services and network; this requires the definition of open and standardized interfaces between each layer, in particular for Control and Service layer in order to allow the third parties to develop and create services independent of the network.
- Broadband end-to-end capabilities; this includes also access network possibilities.
- Support for a wide range of services and applications (real time, non-real time, streaming, transactional, multimedia - unicast, multicast, broadcast) adaptable to varied capabilities of terminals and access networks.
- Global roaming, irrespective of the access mechanism or the technology, i.e. a mobile network user should be able to register in a fixed network as roamer.

The general principle of the NGN architecture is given in figure 1 ("Principe général d'architecture d'un réseau NGN" - see bibliography).

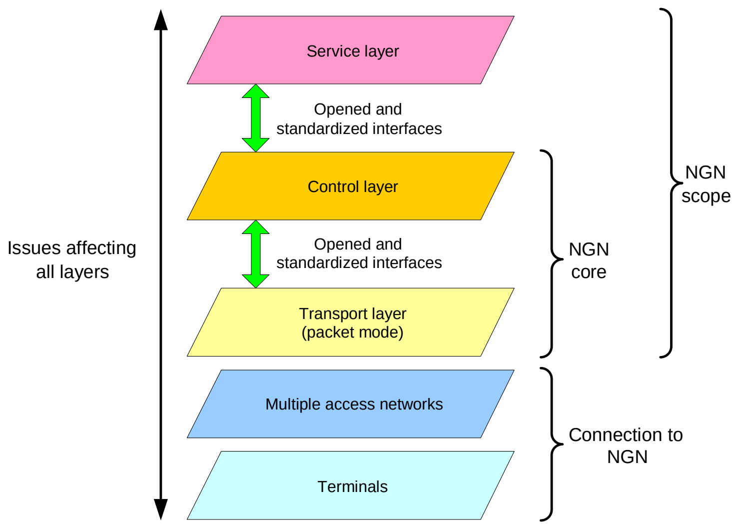

Figure 1: General principle of the NGN architecture

5.2 Accessibility

5.2.1 Broadband access technologies with respect to their use for multimedia services

5.2.1.1 Fixed access

5.2.1.1.1 Analog modems

The analog modem can operate with any dial-up phone. The modem supports high-speed analog data, voice, and fax operation. The ITU-T Recommendation V.90-enabled analog modem supports symmetric rates up to 33,6 kbit/s. In asymmetric mode, it supports data bit rates up to 56 kbit/s from a digitally connected central site modem (downstream). Data can be sent upstream at speeds up to 33,6 kbit/s. The modem can be used for remote access applications such as ISP, online service, or corporate site. However, due to the limited data rate the analog modems are not perspective to be used as an access technology for broadband multimedia services.

5.2.1.1.2 xDSL

The best-known example for broadband access is the Digital Subscriber Line (DSL), a technology that uses the twisted pair for the Plain Old Telephone Service (POTS) in the Local Loop.

The history of DSL started with the concept of Integrated Services Digital Network (ISDN). In the so-called xDSL family, the x in xDSL refers to the number of DSL modem types which include ADSL, SDSL, VDSL, HDSL, and others.

ISDN-BA (DSL) (Integrated Services Digital Network - Basic Access), ADSL (Asymmetric Digital Subscriber Line), RADSL (Rate-Adaptive Digital Subscriber Line), VDSL (Very high bit rate Digital Subscriber Line), HDSL (High bit rate Digital Subscriber Line) and SDSL (Symmetric Digital Subscriber Line) are all DSL modem technologies designed to operate on telephone wires intended originally for voice-band communication (300 Hz to 3,4 kHz).

ISDN-BA connections provide two 64 kbit/s channels plus a 16 kbit/s signalling channel with the possibility to combine the two 64 kbit/s channels into a 128 kbit/s channel. This type of access is used mainly by residential or small users for telephony and/or data. Moreover, the use of two 64 kbit/s channels together gives an opportunity to use it, for instance, for dial-up video services.

ADSL (Asymmetric Digital Subscriber Line)

There are two general categories of DSL: symmetric and asymmetric. Symmetric DSL provides the same service bit-rate in both upstream and downstream direction. Asymmetric DSL (ADSL) provides more downstream bit-rate (from the network to the user) than upstream bit-rate. From the start, ADSL technology was developed to coexist simultaneously with POTS or ISDN voice service. This is achieved by transmitting the data signal at higher frequencies than is used for POTS or ISDN. This type of transmission is referred to as passband transmission. By transmitting a signal at the higher frequencies and avoiding the voice band frequencies, broadband data can be sent simultaneously with voice on the same copper pair. ADSL is now being marketed in most major European countries. ADSL services being commercially offered in Europe can provide a bandwidth of up to 6 Mbit/s downstream and 512 kbit/s upstream, although a maximum downstream speed of 2 Mbit/s is more common. However, the bandwidth that can be offered depends on the distance of the customer from the local exchange. The ADSL is used mainly by residential and small business users as a faster access to the internet.

RADSL (Rate-Adaptive Digital Subscriber Line)

As the name implies, the Rate-Adaptive Digital Subscriber Line modems adjust the data rate to match the quality of the twisted-pair connection. Emerging software should make this an automated process without any difficult human intervention.

VDSL (Very high bit rate Digital Subscriber Line)

VDSL is seen as a technology to be developed for the long term, giving very large bandwidths to customers. Two classes o f payload are considered, i.e. symmetric classes which range from 6,4 Mbit/s up to 28,3 Mbit/s, and asymmetric classes which ranges from 6,4/2 Mbit/s up to 23,2/4 Mbit/s. In theory, a copper pair can carry up to around 50 Mbit/s but it can only do that for short distances - e.g. 300 m for 26 Mbit/s. VDSL technologies and architectures are being developed to make this possible. These will make use of optical fibres for the majority of the access network from the local exchange, with copper pairs (or coax cable TV) being used for the last few hundred meters. The high bandwidth of VDSL and its flexible allocation will also allow different services to be mixed on the same link, e.g. data, video and voice. There are still a number of technical problems to be solved before VDSL becomes a practical solution for widespread application (e.g. cross talk with other pairs in the same cable and radio frequency interference).

HDSL (High bit rate Digital Subscriber Line)

High bit-rate DSL modem standards evolved from earlier work on ISDN-BA. HDSL is a bi-directional symmetric transmission system that allows the transport of signals with a bit-rate of 1,544 Mbit/s or 2,048 Mbit/s on multiple access network wire-pairs. Two different options for the linecode are recommended; the Pulse Amplitude Modulation 2B1Q and the Carrierless Amplitude/Phase modulation (CAP). CAP is applicable for 2,048 Mbit/s, while for 2B1Q two different frames are defined. The 2B1Q standard for 2,048 Mbit/s caters for both duplex transmission on a single pair and parallel transmission on two or three pairs. This allows for the distribution of the data to several pairs and for the reduction of the symbol rate in order to increase the line length or transmission reach. CAP is defined for one or two pairs only and the 1,544 Mbit/s 2B1Q for two pairs only.

SDSL (Symmetric Digital Subscriber Line)

Symmetric or single-pair DSL is very much in the requirements capture phase. It is likely to be symmetric and based on older HDSL technology, but using more advanced techniques to enable greater transmission flexibility over a single wire-pair. SDSL will have application to both business and residential sectors, and could therefore have potentially very high volumes. Data rates range from n x 64 kbit/s to 2,048 Mbit/s in both directions.

SHDSL (Symmetric High-speed Digital Subscriber Line)

SHDSL was based on HDSL and is specified in the ITU recommendation number G.991.2 titled Single-Pair High-Speed Digital Subscriber Line Transceivers. Today SHDSL can operate at data rates from 192 kbit/s to 2,312 Mbit/s (in a 2-wire mode) and 384 kbit/s to 4,624 kbit/s (in a 4-wire mode) with higher rates under development, and is spectrally compatible to all other DSL technologies with the use of TC-PAM line coding. SHDSL combines the best of the legacy services into a single, robust technology that can be used for both full and fractional E1/T1 lines, Digital Added Main Lines (multiple voice channels), and video conferencing applications using a single twisted pair of wires.

An overview of xDSL technologies with respect to their technical parameters and typical usage is given in table 4.

Table 4: Overview of xDSL technologies with respect to the technical parameters and typical usage

Technology	Mode	Standardization	Data rate	Applications
ISDN-BA	symmetric	ITU-T, ETSI	64 kbit/s or 128 kbit/s	Internet access, dial-up video
ADSL/RADSL	asymmetric	ITU-T, ETSI	up to 8 Mbit/s/640 kbit/s (downstream/upstream)	Internet access, VoD, database access, remote LAN access, interactive multimedia
VDSL	asymmetric/ symmetric	ITU-T, ETSI	up to 24/4 Mbit/s (downstream/upstream) up to 36 Mbit/s in a symmetrical mode	same as ADSL plus HDTV
HDSL	symmetric	ITU-T, ETSI	up to 2,048 Mbit/s (E1)	Internet access + symmetric applications (peer-to-peer file sharing, business data traffic), interactive multimedia, replacement of a local repeated E1 trunk
SDSL	symmetric	ITU-T, ETSI	192 kbit/s up to 2,3 Mbit/s	like HDSL
SHDSL	symmetric	ITU-T, ETSI	192 kbit/s up to 2,3 Mbit/s	like HDSL

5.2.1.1.3 Ethernet

Ethernet is the most popular physical layer LAN technology in use today. Other LAN types include Token Ring, Fast Ethernet, FDDI (Fibre Distributed Data Interface), and LocalTalk. Ethernet is popular because it strikes a good balance between speed, cost and ease of installation. These benefits, combined with wide acceptance in the computer marketplace and the ability to support virtually all popular network protocols, make Ethernet an ideal networking technology for most computer users today. The Ethernet standard is defined by the IEEE (Institute for Electrical and Electronic Engineers). The IEEE Standard 802.3 defines rules for configuring an Ethernet network as well as specifying how elements in an Ethernet network interact with one another.

For Ethernet networks that need higher transmission speeds, the **Fast Ethernet** standard has been established. This standard raises the Ethernet speed limit from 10 Mbit/s to 100 Mbit/s with only minimal changes to the existing cable structure. It is a high-speed LAN technology that provides high bandwidth to desktop users, as well as to servers and server clusters.

There are currently two specifications within the IEEE that took different approach with respect to the access method:

- **100BaseT:** It is the IEEE 802.3u specification for the 100 Mbit/s Ethernet implementation over UTP (Unshielded Twisted-Pair and STP (Shielded-Twisted Pair) cabling. The MAC (Media Access Control) layer is compatible with the IEEE 802.3 MAC layer.

- **100VG-AnyLAN:** It is the IEEE specification for 100 Mbit/s Token Ring and Ethernet implementations over 4-pair UTP. The MAC layer is not compatible with the IEEE 802.3 MAC layer. It was developed by Hewlet Packard to support new time-sensitive applications, such as real-time multimedia.

5.2.1.1.4 CATV (Cable Television)

Cable Networks for Television are one of the alternative solutions to provide an access to broadband multimedia services. CATV networks were originally designed for one-way broadcast of television to consumers' homes. To ensure the reception of the cabled TV service with the same TV sets used to receive over-the-air broadcast TV, a portion of the over-the-air radio frequency spectrum within sealed coaxial cables is used. Depending on the technology used within the cable TV networks, the usable bandwidth varies from 450 MHz (all copper coax cable) to 750 MHz (Hybrid Fibre-Coax), composed of channels of 6 MHz or 8 MHz. Each standard television channel occupies 6 MHz of the radio frequency spectrum. It depends on cable operators how they allocate the spectrum for downstream and upstream traffic. Usually the downstream channels are within the frequency range of 50 MHz to 170 MHz and the upstream channels within 5 MHz to 42 MHz.

To deliver data services over a cable network, one of the 6 MHz television channel (in the 88 MHz to 860 MHz range) is allocated for downstream traffic to customers and another channel (in the 5 MHz to 42 MHz range) is used to carry upstream traffic. The transmission rates achieved are between 27 Mbit/s to 36 Mbit/s (depending on the transmission technology), in the downstream, and between 320 kbit/s to 10,24 Mbit/s in the upstream. However, the bandwidth actually available to individual customer might be significantly lower as several customers share the same resource.

The availability of CATV systems is very varied throughout Europe, being very widespread in some countries and relatively uncommon in others. They do give an alternative access network to the copper telephony network with similar standards of reliability and bandwidth. From this point of view, CATV might be a perspective access technology for multimedia interactive services for business and entertainment. The functionality of set-top-boxes and the MHP (Multimedia Home Platform) are integral parts of the broadband achievements.

5.2.1.1.5 PLT (PowerLine Telecommunications)

PowerLine Telecommunications is an access technology that uses the electricity networks with especially prepared signals to provide high speed access to telecommunications services. One major advantage of using the PLT as an access technology is that the power network extends to most rooms in buildings and it therefore provides the potential for communication within the building itself as well as to a local exchange. There are still some open issues like ElectroMagnetic Compatibility (EMC) and the interference of radio systems. Another technical issue that have to be solved involves the frequencies to be used for data transmission. The frequencies needed to allow broadband working are in the same part of the spectrum as the frequencies allocated to police and air traffic control. Power lines are inherently very bad for leaking electromagnetic radiation and the use of these frequencies for broadband transmission could cause serious disturbance to radio services.

5.2.1.1.6 FTTB (Fibre-to-the building)

Fibre-to-the building (FTTB) means a fibre optic running directly to the customer premises. This access technology allows therefore very high bit rates with very high quality.

There are two possibilities how to "bring" fibre to the building:

- All Fibre-PON - intended mainly for residential applications for Internet and other services access.
- All Fibre SDH - intended to provide high-speed, multiservices access to and from business locations.

However, high costs still prevent broad penetration of fibre optic as an access technology.

5.2.1.2 Wireless access

5.2.1.2.1 WLAN (Wireless Local Area Network)

WLAN is a standard offering a limited coverage for LAN users. Cell radius is usually from few tens of meters to some hundred meters. In 1997 IEEE (Institute of Electrical and Electronic Engineering) finished the definition of the WLAN standard 802.11, and in 1999 standards 802.11a and 802.11b [151] have been approved.

IEEE 802.11 [151] wireless LAN is a locally situated network. The coverage area is consisted of small islands (called here BSS - Basic Service Set) and the purpose is certainly not to offer a large coverage network like cellular mobile networks. The coverage area is often tailored according to the users own need and can also be temporary.

The specifications of wireless LAN outline two possible modes of operations: client/server and ad hoc mode WLAN. In the client/server WLAN (sometimes called as an infrastructure configuration) terminals communicate with Base Stations (BS) or Access Points (AP), which form the coverage area. The access points are further connected to the wired network. WLAN access point can comparable to a GSM base station, but in the ad hoc mode the same station acts both as an access point and as a station.

The coverage area of the client/server type WLAN network is usually bordered upon a building or a campus and can therefore be comparable to a single GSM inside cell. The main difference between the GSM and WLAN technologies is that to cover the whole building with the WLAN technology, there has to be several access points depending on the building architecture, wall materials, etc. In the GSM solution the coverage area can be built with a single base station by distributing the signal into antennas locating in different rooms by using power splitters.

The specifications of WLAN define also an ad hoc mode. In this mode mobile terminals by themselves build the network. The coverage area is built by the help of wireless adapters and is limited. In the ad hoc mode the whole network is seen as movable, and it is independent of any infrastructure unlike GSM or client/server type WLAN. It is also isolated, because it has no interface to the wired network.

The first phase of the standard IEEE 802.11 supports only 1 Mbit/s and 2 Mbit/s data rates. The first phase standard was followed by an extension IEEE 802.11b [151] (also referred to as "WiFi"), which operates in the licence free 2,4 GHz band, supports data rates up to 11 Mbit/s and a range up to 300 m.

There are various application fields, like the easy and comfortable setup of an ad hoc network, or the individual mobility of users within a company or home users in their home. Another application is providing Internet access to users in hot spots like airport lounges, hotels, etc.

IEEE 802.11a standard operate at around 5 GHz and enables data rates up to 54 Mbit/s. It is often compared to the ETSI standard HiperLAN2 (see next clause). However, unlike the ETSI HiperLAN/2, the current IEEE 802.11a version does not allow for some features, like power management and dynamic frequency change.

5.2.1.2.2 Radio in the Local Loop

Radio Relay Systems point-to-point or point-to-multipoint or HiperMAN (Metropolitan Area Networks) or HIPERACCESS are fixed radio-based alternatives for broadband access.

ETSI BRAN currently produces specifications for three major standard areas:

- **HiperLAN2**, a mobile broadband short-range access network.
- **HIPERACCESS**, a fixed wireless broadband access network.
- **HIPERMAN**, a fixed wireless access network which operates below 11 GHz.

HiperLAN/2 is a flexible Radio LAN standard which provides high speed access (up to 54 Mbit/s at the physical layer) to different types of networks including ATM and IP based networks, 3G mobile core networks, and also for private use as a wireless LAN system. It operates, similarly like IEEE 802.11a, in the 5 GHz band which has been allocated to wireless LANs worldwide. Basic applications include data, voice and video, with specific QoS parameters taken into account. HIPERLAN/2 systems can be deployed in offices, classrooms, homes, factories, hot spot areas like exhibition halls and more generally where radio transmission is an efficient alternative or a complement to wired technology. The first release of standard has been extended for the home environment. Currently, a work is going on a specification for the access interface to UMTS which could also serve as a basis for the definition of interfaces to the other members of the IMT-2000 family of the 3rd generation mobile systems.

HIPERACCESS is a HiperLAN/2 long variant intended for point-to-multipoint, high speed access (typically, 25 Mbit/s) by residential and small business users to a variety of networks including ATM and IP based networks and UMTS core networks. It is targeting high frequency bands, spectrum allocation in the 40,5 GHz to 43,5 GHz are being discussed. The range is up to 5 km. As an access technology, it can be seen as a wireless alternative to fixed line high-bandwidth access networks implemented with fibre cables.

HIPERMAN is intended to operate at radio frequencies between 2 GHz and 11 GHz. The air interface will be optimized for PMP (Point-to-MultiPoint) configurations, but may allow for flexible mesh deployments. The HIPERMAN standards will specify the physical and link layers, which are core network independent, and the core network specific convergence sublayers. In order to specify a complete system, other specifications, e.g. for the network layer and higher layers are required. These specifications are assumed to be available or to be developed by other bodies.

5.2.1.2.3 Bluetooth

Bluetooth is the name given to a technology standard using short-range radio links, intended to replace the cable(s) connecting portable and/or fixed electronic devices. The standard defines a uniform structure for a wide range of devices to communicate with each other. Its key features are robustness, low complexity, low power and low cost, which make it especially suited to mobile handheld devices. The technology also offers wireless access to LANs.

PSTN, the mobile phone network and the Internet for a host of home appliances and portable handheld interfaces.

Bluetooth enabled electronic devices connect and communicate wirelessly via short-range, ad hoc networks called piconets. Each unit can simultaneously communicate with up to seven other units per piconet. Moreover, each unit can simultaneously belong to several piconets. These piconets are established dynamically and automatically as Bluetooth devices enter and leave the radio proximity.

The theoretical bit rate is 1 Mbit/s. Bluetooth can support:

- an asynchronous data channel (asymmetric link of maximally 723 kbit/s in the forward direction and 57 kbit/s in the return direction, or symmetric link with the bit rate up to 434 kbit/s);
- up to three simultaneous synchronous voice channels (each voice channel supports 64 kbit/s synchronous voice link in each direction);
- a channel which simultaneously supports asynchronous data and synchronous voice.

Bluetooth's main strength is its ability to simultaneously handle both data and voice transmissions, allowing such solutions as a mobile hands-free headset for voice calls, print to fax capability, and automatically synchronizing PDA, laptop, and cell phone address book applications. The future versions of Bluetooth are supposed to support video and streaming multimedia.

5.2.1.2.4 DECT

DECT (Digital Enhanced Cordless Telecommunications) is a flexible digital radio access technology for cordless communications in residential, business and public environments. Designed for short-range use (20 m to 300 m) as an access mechanism to the main networks, DECT offers cordless voice, fax, high speed data and multimedia communications, wireless local area networks and wireless PBX. It thus has applications in the factory, office, home and public areas. For network operators DECT is also a cost-effective and flexible alternative to conventional cable/fibre connections into customers' premises, by providing "Wireless Local Loop". Frequency bands have been made available for DECT in more than 100 countries worldwide. DECT services are compatible with GSM and ISDN and dual-mode DECT/GSM handsets are available. In a large number of countries DECT operates in a protected band, i.e. no interference from other technologies. ETSI DECT is one of the technologies the IMT-2000 specification.

The ETSI DECT norm has been enhanced for the support of:

- DECT Packet Radio Service (DPRS), which introduces flexibility and resources optimization of packet technology, and opens the field to further developments in terms of speed and profiles. By adding higher bit rate modes to DECT, DPRS can currently support data applications up to more than 2 Mbit/s. The third version of DPRS should support the transmission of broadband data up to more than 10 Mbit/s.
- Application Specific Access Profiles (ASAPs) is a group of industry-driven standards that have been created in order to enhance interoperability. Each of them identifies a specific application scenario and selects a specific subset of DPRS services plus a voice service for particular applications.
- DMAP (DECT Multimedia Access Profile) has been developed for residential/SOHO applications. It allows design of low-cost domestic devices for local data interconnection and Internet connection via PSTN/ISDN. Ethernet (Eth) Interworking and V.24 Interworking were also defined. They define data ASAP combining a selection of Ethernet and V.24 Interworking DPRS data services.

DECT supports also mobile internet technology, specifying the additional requirements for DECT IP applications. The work is now focusing on the enhancement of the system's specifications with respect to cover high bit rates and multimedia applications.

5.2.1.3 Cellular systems

5.2.1.3.1 GPRS (General Packet Radio Service)

One major difference between "wireless" and "cellular" access is a range of mobility provided to a user. While a wireless access itself provides a user only with a very limited mobility within the range of this access point, full mobility can only be achieved by an underlying cellular network, which implements the mobility across the whole area covered by the mobile cellular network. After several regional variants of mobile networks, the GSM (Global System for Mobile Communications) was introduced and accepted in the early nineties. The GSM standard uses the radio spectrum around 900 MHz, 1 800 MHz and 1 900 MHz in a licensed band. Due to the scarcity of bandwidth, the GSM standard allows only 13 kbit/s for speech transmission, which leads to a noticeable reduction of the voice quality in GSM phone calls compared to fixed line telephone. For the data transmission, GSM is limited to 9,6 kbit/s only. As this data rate is too slow for most of the current data services, the GPRS (General Packet Radio Service) was defined as an extension to the GSM standard. Though the GPRS standard allows data rates theoretically up to 171 kbit/s, current networks and terminals allow not more than 50 kbit/s for downstream and 13,4 kbit/s for upstream.

5.2.1.3.2 EDGE (Enhanced Data for GSM Evolution)

EDGE is another extension to GSM networks that was standardized to further increase the data rate in mobile networks. By an enhancement of the radio interface, EDGE is capable of data rates up to 384 kbit/s.

Both the GPRS and EDGE require upgrade of the existing GSM networks, and both also require a new type of terminal for a user who wishes to use the technologies. Unlike the UMTS, no new licence is required for GPRS and EDGE providers. It is a reason why the both technologies might be the solution for those network operators that were not successful in achieving a licence for UMTS. They might be access technologies of choice to high speed services for those users who cannot benefit from the immediate availability of UMTS.

5.2.1.3.3 UMTS (Universal Mobile Telecommunication Services)

UMTS is one of the major new third generation (3G) mobile systems being developed within the framework which has been defined by the ITU (International Telecommunications Union) and known as IMT-2000 (International Mobile Telecommunications). UMTS will play a key role in creating the future mass market for high-quality mobile multimedia communications. UMTS seeks to build on and extend the capability of today's mobile, cordless and satellite technologies by providing increased capacity, data capability and a far greater range of services using an innovative radio access scheme and an enhanced, evolving core network.

UMTS is being standardized by the European Telecommunications Standards Institute (ETSI) in the IMT-2000 framework, in co-operation with other regional and national standardization bodies around the world. IMT-2000 has been defined by the ITU as an open international standard for a high capacity, high data rate mobile telecommunications system incorporating both terrestrial radio and satellite components. UMTS is an important part of wider initiatives to satisfy the needs of corporate users and the mass market. Complementary work is under way throughout ETSI and other fora on every aspect of the emerging information society, multimedia, information and content.

In December 1998, a body called the 3rd Generation Partnership Project (3GPP) was established with the aim of harmonizing the various proposals (based on W-CDMA) submitted by various countries or regions for the multiple access schemes to be employed on the air interface. It was founded by the following regional standardization bodies: ARIB (Japan), TTC (Japan), ETSI (Europe), T1 (US), TTA (Korea), and joined later by CWTS (China). Subsequent to the establishment of 3GPP, a second body, 3GPP2, was established around cdma2000 proposal. In June 1999, a group of international operators, the Operator Harmonization Group (OHG), proposed the harmonization of 3GPP and 3GPP2 concepts, to be known as Global Third Generation (G3G), in order to allow interoperability between UTRA and cdma2000. The proposals of OHG were accepted by both 3GPP and 3GPP2 to produce a standard with the following three modes of operation:

- CDMA-DS (Code Division Multiple Access - Direct Sequence), based on UTRA FDD.
- CDMA-TDD, based on UTRA TDD.
- CDMA-MC (CDMA Multi Carrier) based on cdma2000.

In November 1999, five models have been adopted by ITU as IMT-2000 standards:

- IMT-MC (multicarrier): standard cdma2000.
- IMT-SC (single carrier): standard UWC-136.
- IMT-DS (direct sequence): standard UTRA FDD.
- IMT-TC (TDMA/CDMA): standards UTRA TDD, TD-SCDMA.
- IMT-FT (FDMA/TDMA): standard DECT.

The key difference between this system and previous mobile systems, such as GSM is that the earlier systems were conceptually separate from the fixed telephone network. The UMTS integrates fixed (wired) and wireless systems to provide a universal communications service, such that a user can move from place to place while maintaining access to the sum set of services. From this point of view, the UMTS can be considered an accelerator for tomorrow's wireless information society, delivering high-value broadband information, commerce and entertainment services to mobile users via fixed, wireless and satellite networks.

An overview of wireless and cellular technologies with respect to their technical parameters and typical usage is given in table 5 (based on "Wireless Access Tutorial"- see bibliography).

Table 5: Overview of wireless and cellular technologies with respect to the technical parameters and typical usage

Technology	Standardization	Frequency	Data rate	Range	Application
WLAN 802.11b	IEEE	2,4 GHz	11 Mbit/s	150 m	WLAN
WLAN 802.11b	IEEE	5,15 GHz	54 Mbit/s	150 m	WLAN
HiperLAN/2	ETSI EP BRAN	5,2 GHz (in 4 GHz and 5 GHz band licensed)	54 Mbit/s	30 m to 200 m	WLAN, local access to ATM
HIPERACCESS	ETSI EP BRAN	40,5 GHz/43,5 GHz	25 Mbit/s	5 km	remote access to IP/ATM/UMTS
Bluetooth	Bluetooth SIG	2,4 GHz	721 kbit/s	0,1 m to 10 m to 100 m	periphery devices
DECT	ETSI EP DECT	1,9 GHz	552 kbit/s, up to 2 Mbit/s	300 m	corporate networks, for operators: complementary to 3G mobile systems
GPRS	ETSI TS SMG	0,9 GHz/1,8 GHz/ 1,9 GHz	171 kbit/s	up to 30 km to 50 km to the next BS, less for max. data rates	data services, WAP
EDGE	3GPP GERAN	0,9 GHz/1,8 GHz/ 1,9 GHz	384 kbit/s	as well as for GPRS	data services, 3G systems in the U.S.
UMTS	3GPP	1,90 GHz to 1,98 GHz, 2,01 GHz to 2,025 GHz, 2,11 GHz to 2,17 GHz	144 kbit/s, 384 kbit/s, 2 Mbit/s	depending on number of users in cell	voice, data, multimedia services

5.2.1.4 Terrestrial systems

These are the systems, which use radio links to provide connections between customers in fixed locations and telecommunications networks. It covers systems such as LMDS (Local Microwave Distribution System), MMDS (Multichannel Multipoint Distribution Service), MVDS (Microwave Video Distribution Systems) and MWS (Multimedia Wireless Systems). They mostly work in the 25 GHz or 40 GHz area of the radio spectrum.

These systems are well suited to broadcast and multi-cast applications but also provide broadband data links to and from the customer. Compared to a satellite, individual terrestrial wireless systems have a much smaller coverage area (e.g. a radius of 5 km). This means that their radio spectrum can be reused many times across a geographical area. For residential users, IP would normally be used over the system, but ATM connections may be offered to some business users.

Terrestrial wireless systems should provide a similar level of reliability to ADSL systems, but does need line of sight communication.

MMDS (Multichannel Multipoint Distribution Service) uses microwave transmission to provide Internet access downlinks. When combined with telephony uplinks, it provides a complete Internet access. Typical data rates for a LMDS system would be 36 Mbit/s (shared) downstream and 8 Mbit/s upstream.

LMDS (Local Multipoint Distribution Service) is similar to the MMDS in that it will use microwave transmission for Internet access downlinks and telephony links for uplinks. It will work in higher frequency band (above 20 GHz depending on country of licensing) and will have about four times the bandwidth of the MMDS. These facts, together with its relatively close transmitter spacing, should enable LMDS to serve much higher density of Internet users than MMDS. It can be used for a variety of applications such as digital two-way voice, data, high-speed Internet access and video services.

5.2.1.5 Satellite systems

Several approaches have been proposed for using satellites to provide two-way data services to users around the world. Published expected data rates of the various satellite constellations vary between 200 kbit/s downstream to 2 Mbit/s upstream for residential users and from 10 Mbit/s to 30 Mbit/s for business users. Satellite systems are well suited for providing access to areas which are sparsely populated and which lack other alternatives.

5.2.1.5.1 Low Earth Orbit (LEO) satellites

Communications satellites have until recently been mainly geostationary, i.e. they appear to hover above a fixed point on the earth. This has the major advantage that the antenna used to connect the ground station to the satellite does not have to track the path of the satellite across the sky and calls do not have to be handed over from one satellite to the next as the first satellite flies below the horizon.

The major disadvantage of a geostationary satellite is that it has to be at around 36 000 km above the earth. This means that the signal being received from the satellite is not very strong (and that any transmitter from the ground to the satellite has to provide a strong signal) and that the transmission time to and from the satellite is high. Anybody who has made a phone call via a satellite will be aware of the slightly disconcerting effects of this. This delay can also affect protocols which assume failure if they do not receive a rapid response.

One solution is to provide satellites in a much lower orbit of around 1 000 km. These have the advantage of a much lower delay and the reduced power needed means it is much easier to design aerials for bi-directional communication. By using LEO satellites it is possible to provide users with a high bandwidth for download and upload. The main disadvantage of LEO systems is that a large number of satellites (between 50 and 300) are needed for good coverage of the world.

LEO satellites have the disadvantage that they move relative to the earth. This means that they will spend a percentage of their time over parts of the earth's surface where there is little demand for them and, conversely, a large number of satellites are needed to provide constant service to any one location.

5.2.1.5.2 High Altitude Platform Stations (HAPS)

An alternative approach, which is being considered, is High Altitude Platform Stations. There are a number of variations of these being proposed, but a typical scheme is proposing a balloon-based platform about the size of a football field at a height of 21 km. The plan is to launch these over large cities. These would offer broadband communication similar to that which is possible from satellite, with simpler aerial design and transmission times similar to the terrestrial network. The main problems that have to be resolved for HAPS are with the physical platform rather than with the communications equipment.

5.2.1.6 Broadcast systems

Both the terrestrial and satellite broadcast systems were originally set up to distribute radio and TV programs, mainly to residential sites. Whilst the analogue services were the only ones available, this remained the only use of this network apart from the text pages broadcast by teletext. The introduction of digital broadcast services (Digital Video Broadcast and Digital Audio Broadcast) has opened up new opportunities for interactivity. Many broadcasters are now offering services which use the digital terrestrial broadcast system for downstream transmission and use a connection through the telephone network for the upstream connection. There are limitations with this system, such as the need to share the downstream bandwidth between a large number of users. The services, which can be offered are very similar to those provided by satellite.

In Europe, the Digital Audio Video Council (DAVIC) issues the Digital Video Broadcast (DVB) [5] specifications for digital set-tops and for cable modems. The DVB/DAVIC cable modem specifications, "EuroModem", are an alternative to DOCSIS (Data Over Cable Service Interface Specification - the international cable modem standard, DOCSIS 1.1 - ITU-T Recommendation J.112 [6], DOCSIS 2.0 - ITU-T Recommendation J.122 [7] and were established to better fit the European set-top devices. A similar entity to CableLabs, the EuroCableLabs (ECL) operates under the direction of the European Cable Communications Association (ECCA).

The DVB 2.0 specifications, adopted by the ETSI [8], were also selected by DAVIC as the DAVIC 1.5 specifications for cable modems. They describe the out-of-band and in-band transmission options applicable to interactive set-top boxes and cable modems for the deployment of interactive TV, data and voice services over a common cable network platform.

DVB, via terrestrial and satellite broadcasting, could implement some of broadband multimedia services for fixed, but also mobile users.

DVB-S systems are used extensively in Direct To Home (DTH) television broadcasting, and new satellite systems are being deployed to support digital audio broadcasting services.

DVB-T in the VHF and UHF bands may be suitable for such applications as data, speech, Internet services as well as TV in the MPEG-2 transport stream.

DVB Return Channel via Satellite (DVB-RCS), recently published as an ETSI standard [9], forms the specification for the provision of the interaction channel for GEO satellite interactive networks with fixed Return Channel Satellite Terminals (RCST). The standard specifies a satellite terminal (sometimes known as a Satellite Interactive Terminal (SIT) or Return Channel Satellite Terminal (RCST)) supporting a two-way DVB satellite system.

5.2.1.7 Conclusions

Demand for high data rates has lead to several technological approaches developed to provide broadband access to business and residential customers. The preceding clause has described some of the technologies, which are available, or are becoming available, to access broadband multimedia services.

There are a number of trends that can be seen in the evolution of the access technologies:

- There is a big diversity in the technologies used for access networks.
- There is the continued high growth in demand for Internet-based services. For most residential and small business users, that is likely to be provided by ADSL or cable modems.
- An alternative view of evolution for Internet access is that many residential customers will not own PCs in near future, but they will still willingly pay for TVs. From this point of view, interactive services via TV will become more important, including the provision of e-mail and e-commerce services. The evolution of the satellite, cable and terrestrial broadcast access networks can play a key role. The ability of these networks to interwork with telecommunications networks (for the return path) can be crucial, at least in the early stages.
- Mobility is increasingly important for users. Mobile phones and networks are becoming more capable and can already offer basic information services and e-mails over the Internet. With the advent of GPRS and, later, UMTS they will become much better at handling data-based traffic, such as Internet access. However, the radio spectrum available is always going to be much more limited than the bandwidth that is available over land lines. This means that the fixed network is always going to be able to provide much higher bandwidth service than mobile networks. For those reasons, the fixed network will probably be the major provider of broadband multi-service facilities.

5.3 Transport layer

The transport layer of the core network has to be designed in such a way to be able to support any type of service or application with differing sensitivities to delay and with different variations in traffic loading. At the physical level there is a basic topology of physical means (wired or wireless) that interconnects the nodes of the network. There are several methods of sharing and interconnecting the physical capacity. The objective of the design is to maximize the traffic handling capacity within the quality constraints of the services being supported.

- At the physical level, the SDH (Synchronous Digital Hierarchy) system is used to subdivide the capacity on the cable fibres and to add some basic fault monitoring capability. At nodes, cross connects are used to interconnect different units of capacity on different cables. However, SDH is not essential and some companies are developing "IP over λ" routers that put IP packets directly onto optical modulators.

- At the link layer, a packet type system is used to create a more finely grained system of links to time share capacity on the topology created by SDH. This is either the ATM (Asynchronous Transfer Mode), or the FR (Frame Relay). The ATM links connect ATM routers that switch ATM cells in accordance with a link identifier, which is a combination of VP/VC (Virtual Path/Virtual Channel) Identifiers.

- At the network layer, traffic between end users is sent in IP packets that are routed by IP routers at the network layer nodes according to the IP address of the destination.

- IP packets can run on ATM that in turn runs on SDH or run directly on SDH, i.e. the link layer can be null.

- IP networks always have IP routers at their edges or access points to route incoming and outgoing packets. Inside the network, however, either other IP routers can be placed at the nodes or ATM routers can be used to create a mesh of virtual connections between the IP routers at the edge. Although the use of ATM adds additional overheads in terms of the ATM headers, it has several advantages (e.g. ATM routers introduce less delay than IP routers, they also offer more flexibility in running multiple queues; ATM can distinguish between 5 different quality classes, whereas mechanisms for distinguishing quality in IP are not yet fully developed). These advantages are especially valuable for conversational voice, which is highly sensitive to delay.

There is a further development called MPLS (Multi-Protocol Labelling System) [10]. With MPLS, IP routers at the edges of a network attach locally defined labels to IP packets. These labels define a FEC (Forwarding Equivalence Class) and they are used by internal IP routers for routeing (label edge routers can use labels to define a whole route through a complex network of ATM nodes) and for distinguishing different classes of traffic for quality differentiation.

The use of labels reduces the processing load on the router because the label can be analysed more easily and quickly than the IP address. The main advantages of adding label switching to an IP core network that does not use ATM are the reduction in delay, the addition of source route control and the introduction of different quality classes.

Where the internal routers are ATM or FR rather than IP, the label is carried in the ATM header instead of the VPI/VCI (Virtual Path/Virtual Channel Identifier). The three main advantage of adding label switching to ATM are that:

- MPLS has a well defined protocol for establishing labels for routes within a network.

- MPLS labels can be used with IP, ATM and FR and so can work across a network of mixed technologies.

- MPLS is better equipped for rapid reconfiguration for congestion control.

For IP packets that relate to signalling and data, there are four options inside a network:

- IP alone.

- IP with MPLS.

- IP on ATM.

- IP with MPLS on ATM.

For media, there is the additional option not to use IP at all but to run the media directly over ATM using the ATM Adaptation Layer AAL1 or AAL2. This is the lowest delay solution. Since the media stream in applications such as telephony, or videoconferencing is much more sensitive to delay than the signalling, the option of running the signalling as IP on ATM with the media going directly on ATM is quite attractive.

The design and management of IP/ATM networks is the subject of much development work at present.

5.4 Signalling and protocols (Control layer)

Two of the principal characteristics of the NGN given in clause 5.1:

- it is be based on a unique and shared packet-based network (IP, ATM, etc.);
- transport layer and control layer are independent.

This brings also the necessity of the separation of control functions for bearer control, call/session control and service/application control functions and leads to:

- replacement of the traditional switching systems by new types of functional elements at the control layer;
- appearance of new signalling protocols between different functional elements.

5.4.1 Functional elements

This clause gives an overview of the control layer functional architecture. The full NGN architecture comprises network elements needed for the provision of traditional telephony services and advanced next-generation applications (Eurescom Project P1109: Next Generation Networks: the service offering standpoint - see bibliography).

5.4.1.1 Media Gateway

The Media Gateway converts media and framing protocols provided in one type of network to the format required in another type of network. The Media Gateway terminates the bearer control protocols and contains bearer terminations. It also contains media manipulation equipment (e.g. transcoders, echo cancellers, or tone senders).

The MG should also provide a Signalling Gateway function when it terminates call signalling from access networks and has to relay it to the Call/Session Server. The Media Gateway may therefore provide terminations such as Q.931 on the PSTN's side and IUA (ISDN User Adaptation layer) over SCTP/IP on the NGN side.

The Media stream policy/classification functions (e.g. control of traffic contracts: peak cell rate, average cell rate etc. in ATM, QoS class/priority management for Diffserv/RSVP in IP) are also provided by the Media Gateway.

5.4.1.2 Signalling Gateway

The Signalling Gateway is the network entity responsible to forward call control signalling by converting the transport mechanism of the incoming signalling to an appropriate ongoing transport mechanism (e.g. SS7 call signalling over MTP onto SS7 over IP).

In order to provide this function the Signalling Gateway may exist as a separate physical entity or reside within the same physical entity that support Media Gateway function depending on the particular network scenario. In the case that the Call/Session server terminates call signalling from access network, the Signalling Gateway function is useless.

5.4.1.3 Call/Session Server (Media Gateway Controller)

A Call/Session Server provides basic call control including call routing (routing tables, address translations between different numbering plan formats, routing information retrieval from external devices), call signalling process (SIP, H.323, ISUP, MGCP, etc.) and H.248-like Media Gateway Controller functions.

It should also provide more advanced call control functions like third party call control and CLASS services.

In addition, it must provide standard and open interfaces towards Application Servers to enable call related event triggering, service and policy control (e.g. personalized QoS policies, AAA policies, etc.).

In the NGN architecture, the Call/Session Server presents the central node that supports the intelligence of communication.

The Call/Session Server may exist as a separate physical entity or reside within the same physical entity that supports Media Gateway function depending on the particular network scenario.

Multiple Call/Session Servers may co-operate in order to handle a single call thank to server-to-server protocols such as SIP-T or BICC.

Examples of Call/Session servers are call agents, softswitches, SIP Server and H.323 gatekeepers.

5.4.1.4 Physical implementation

Thanks to decomposition of the network functions into layers the physical implementation and geographical localization of the functional elements mentioned above is left on the choice of constructors and network operators. This also allows to optimize the network resources and to dimension them separately.

5.4.2 Candidate protocols

5.4.2.1 Call control protocols

Call control protocols allow the establishment, in general based on the user demand, the communication between two terminals or between a terminal and a server. The two candidate protocols are H.323 and SIP.

5.4.2.1.1 H.323

ITU-T Recommendation H.323 is a set of protocols for voice, video, and data conferencing over packet-based networks such as the Internet. The H.323 protocol stack is designed to operate above the transport layer of the underlying network. As such, H.323 can be used on top of any packet-based network transport like Ethernet, TCP/UDP/IP, ATM, and Frame Relay to provide real-time multimedia communication. H.323 uses the IP for inter-network conferencing.

H.323 is one of several videoconferencing recommendations issued by ITU-T.

The other recommendations in the series include H.310 for conferencing over broadband ISDN (B-ISDN), H.320 for conferencing over narrowband ISDN, H.321 for conferencing over ATM, H.322 for conferencing over LANs that provide a guaranteed quality of service, and H.324. The H.324 standard is a system specification for multimedia terminals operating on circuit-switched networks including: terminals on the PSTN ("H.324/P"), terminals on wireless networks (annex C of H.324, "H.324/M"), and terminals on the ISDN (annex D of H.324, "H.324/I").

H.323 architecture:

The recommendation is based on the definition of the components, protocols, and procedures defined in H.323. H.323 architecture is given in figure 2.

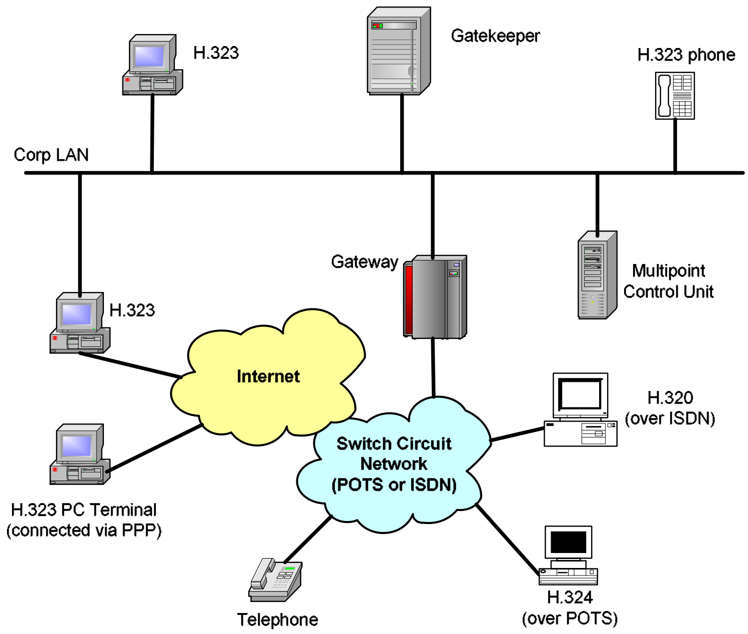

Figure 2: H.323 architecture

H.323 components:

H.323 Terminal: A terminal, or a client, is an endpoint where H.323 data streams and signalling originate and terminate. It may be a multimedia PC with a H.323 compliant stack or a standalone device such as IP telephone. A terminal must support audio communication; video and data communication support is optional.

Gateway: A gateway is an optional component in a H.323-enabled network. However, when communication is required between different networks a gateway is needed at the interface. Through the provision of gateways in H.323 it is possible for H.323 terminals to inter-operate with other H.32X compliant conferencing terminals. A gateway provides data format translation, control signalling translation, audio and video codec translation, and call setup and termination functionality on both sides of the network. Depending on the type of network to which translation is required a gateway may support H.310, H.320, H.321, H.322, or H.324 endpoints.

Gatekeeper: A gatekeeper is an optional component of an H.323-enabled network. Gatekeepers are needed to ensure reliable, commercially feasible communications. A gatekeeper is often referred to as the "brain" of the H.323 enabled network because of the central management and control services it provides. When a gatekeeper exists all endpoints (terminals, gateways, and MCUs) must be registered with it. Registered endpoints' control messages are routed through the gatekeeper. Basic functions provided by a gatekeeper are address translation, admission and access control of endpoints, bandwidth management and routing capability.

MCU (Multipoint Control Unit): A multipoint control unit enables conferencing between three or more endpoints. Although the MCU is a separate logical unit it may be combined into a terminal, gateway, or gatekeeper. The MCU is an optional component of an H.323-enabled network.

Call control and signalling protocols and procedures:

The flow of information in a H.323-enabled network consists of a mix of audio, video, data, and control packets. Control information is essential for call setup and tear down, capability exchange and negotiation, and administrative purposes. H.323 uses three control protocols: H.245 media control, H.225/Q.931 call signalling, and H.225.0 RAS:

H.225.0 Call Signalling: Call signalling is a basic requirement needed to set up and tear down a call between two endpoints. H.225.0 uses a subset of Q.931signalling protocol for this purpose. H.225.0 adopts Q.931 signalling by incorporating it in its message format. H.225.0 call signalling is sent directly between the endpoints when no gatekeeper exists. When a gatekeeper exists then it may be routed through the gatekeeper.

H.245 Media Control: The flexibility of H.323 requires that endpoints negotiate to determine compatible settings before audio, video, and/or data communication links can be established. H.245 uses control messages and commands that are exchanged during the call to inform and instruct. The implementation of H.245 control is mandatory in all endpoints. H.245 control messages may also be routed through a gatekeeper if one exists. H.245 provides the following media control functionalities:

- Capability exchange: H.323 allows endpoints to have different receive and send capabilities. Each endpoint records its receiving and sending capabilities (e.g. media types, codecs, bit rates, etc) in a message and sends it to the other endpoint(s).

- Opening and closing of logical channels: H.323 audio and video logical channels are uni-directional end-to-end links (or multipoint links in the case of multipoint conferencing). Data channels are bi-directional. A separate channel is needed for audio, video, and data communication. H.245 messages control the opening and closing of such channels.

- Flow control messages: These messages provide feedback to the endpoints when communication problems are encountered.

- Other commands and messages: Several other commands and messages may be used during a call like a command to set the codec at the receiving endpoint when the sending endpoint switches its codec.

H.225.0 RAS (Registration, Admission, Status) messages define communications between endpoints and a gatekeeper. H.225.0 RAS is only needed when a gatekeeper exists. Unlike H.225.0 call signalling and H.245, H.225.0 RAS uses unreliable transport for delivery. In an IP network H.225.0 RAS uses UDP.

H.323 scope:

The scope of H.323 does not include the audio/video capture equipment. It is assumed that the audio and video digital streams are available to the H.323 terminal for processing. The Real-time Transport Protocol (RTP) and the associated control protocol - Real-time Control Protocol (RTCP) - are employed for timely and orderly delivery of audio and video streams. RTP/RTCP is an Internet Engineering Task Force (IETF) recommendation that provides logical framing, sequence numbering, timestamping, payload distinction (e.g. between audio and video and between different codecs), and source identification. It may also provide basic error detection and correction. Note that the RTP layer is above the transport layer of the underlying network.

Addressing:

Each H.323 entity shall have at least one Network Address. This address uniquely identifies the H.323 entity on the network. Some entities may share a Network Address (i.e. a terminal and a co-located MC). This address is specific to the network environment in which the endpoint is located. Different network environments may have different Network Address formats. An endpoint may use different Network addresses for different channels within the same call.

For each Network Address, each H.323 entity may have several TSAP (Transport layer Service Access Point) Identifiers. These TSAP Identifiers allow multiplexing of several channels sharing the same Network Address.

An endpoint may also have one or more alias addresses associated with it. An alias address may represent the endpoint or it may represent conferences that the endpoint is hosting. The alias addresses provide an alternate method of addressing the endpoint. These address include dialedDigits or partyNumber addresses (including private telephone numbers and public E.164 numbers), H.323 IDs (alphanumeric string representing names, e-mail like addresses, etc.), and any others defined in the ITU-T Recommendation H.225.0.

H.323 version 2:

Approved in January of 1998, version 2 of the H.323 standard addresses many deficiencies in version 1 and introduces new functionality within existing protocols, such as H.245 and H.225, as well as new protocols. The high-level overview of the most important changes follows:

- **Fast Connect:** is a new method of call setup that bypasses some usual steps in order to make it faster. In addition to the speed improvement, Fast Connect allows the media channels to be operational before the CONNECT message is sent, which is a requirement for certain billing procedures.

- **Supplementary Services for H.323:** namely Call Transfer and Call Diversion, have been defined by the H.450 series. H.450.1 defines the signalling protocol between H.323 endpoints for the control of supplementary services. H.450.2 defines Call Transfer and H.450.3 Call Diversion. The hooks for Supplementary Services are specified in H.323 Version 2. The proper usage of these hooks is specified in H.450.x.

- **QoS structures:** have been added to the H.245 OLC (Open Logical Channels) packets to allow endpoints to set QoS parameters for the media streams, including RSVP (Resource Reservation Protocol) parameters. H.323 only communicates QoS information between H.323 devices. Actual reservation and control of resources is outside the scope of the standard.

- **New types of alias addresses:** version 2 adds support for four additional alias types: Email, URL, Transport ID and Party Number.

H.323 version 3 and 4:

H.323 version 3 was approved on September 30, 1999. H.323 version 3 makes modest improvements to the Recommendation H.323 version 2, introducing only a few new features to the base document. However, H.323 has progressed substantially, mostly in the form of new annexes to H.323 and H.225.0 that add considerable value to the overall H.323 system architecture. Several new supplementary service documents have been added to the H.323 series, including H.450.4 (Call Hold), H.450.5 (Call Park and Pickup), H.450.6 (Message Waiting Indication), and H.450.7 (Call Waiting).

Version 4 was approved on November 17, 2000 and contains enhancements in a number of important areas, including reliability, scalability, and flexibility. New features help facilitate more scalable Gateway and MCU solutions to meet the growing market requirements. The most important new features are:

- **Gateway decomposition:** recognizing the need to build larger, more scalable gateway solutions for carrier solutions, the ITU-T Recommendation SG 16, together with the IETF, produced the new Recommendation H.248, which describes the protocol between the Media Gateway Controller (MGC) and the Media Gateway (MG). To support this "decomposition" of the Gateway, H.323 contains a new clause that describes some of the various architectural designs that may be achieved by decomposing the Gateway into the separate MGC and MG.

- **Supplementary services:** H.450.8 (Name Identification Service), H.450.9 (Call Completion), H.450.10 (Call Offer), H.450.11 (Call Intrusion).

- **Annex K/H.323:** describes a means of providing HTTP-based control for H.323 devices. With this annex, service providers have the ability to display web pages to the user with meaningful content that ties into the H.323 systems. In fact, it is a third party call control mechanism that utilizes a separate HTTP connection for control.

- **Annex L/H.323:** provides a new "stimulus-based" control mechanism for H.323. With annex L, an H.323 device may communicate with a feature server to provide the user with various services. The H.323 endpoint may possess some intelligence, but some intelligence may reside only in the feature server or multiple feature servers.

ITU-T Recommendation SG 16 is currently working on H.323, version 5.

5.4.2.1.2 SIP (Session Initiation Protocol)

The Session Initiation Protocol, or SIP, is an IETF text-based signalling protocol [11] for establishing real-time multimedia calls and conferences over Internet Protocol networks. Each session may include different types of data such as audio and video, although currently most of the SIP extensions address audio communication. As a traditional text-based Internet protocol, it resembles the Hypertext Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP). SIP uses Session Description Protocol (SDP) for media description. SIP is transport-layer independent because it can be used with any datagram or stream protocol (UDP, TCP, ATM, etc.). SIP is text-based in that a method is formed via a textual header that has fields that contain call properties. This text-based approach is easy to parse, thin in terms of packet overhead and very flexible.

SIP components

SIP's basic architecture is client/server in nature. The main entities in SIP are the User Agent, SIP network servers and the Registrar:

- **User Agent**: The User Agents, or SIP endpoints, function as clients (UACs) when initiating requests and as servers (UASs) when responding to requests. User Agents communicate with other User Agents directly or via an intermediate server. The User Agent also stores and manages call states.

- **SIP network servers**: SIP network servers have the capability to behave as proxy or redirect servers.

  - **SIP Proxy Servers**: They forward requests from the User Agent to the next SIP server, User Agent within the network and also retain information for billing/accounting purposes.

  - **SIP Redirect Servers**: They respond to client requests and inform them of the requested server's address. Numerous hops can take place until reaching the final destination. SIP's tremendous flexibility allows the servers to contact external location servers to determine user or routing policies, and therefore, does not bind the user into only one scheme to locate users. In addition, to maintain scalability, the SIP servers can either maintain state information or forward requests in a stateless fashion.

- **SIP Registrar**: The third entity that comprises SIP is the SIP Registrar. The User Agent sends a registration message to the SIP Registrar and the Registrar stores the registration information in a location service via a non-SIP protocol. Once the information is stored, the Registrar sends the appropriate response back to the user agent.

The interaction between SIP components is depicted in figure 3.

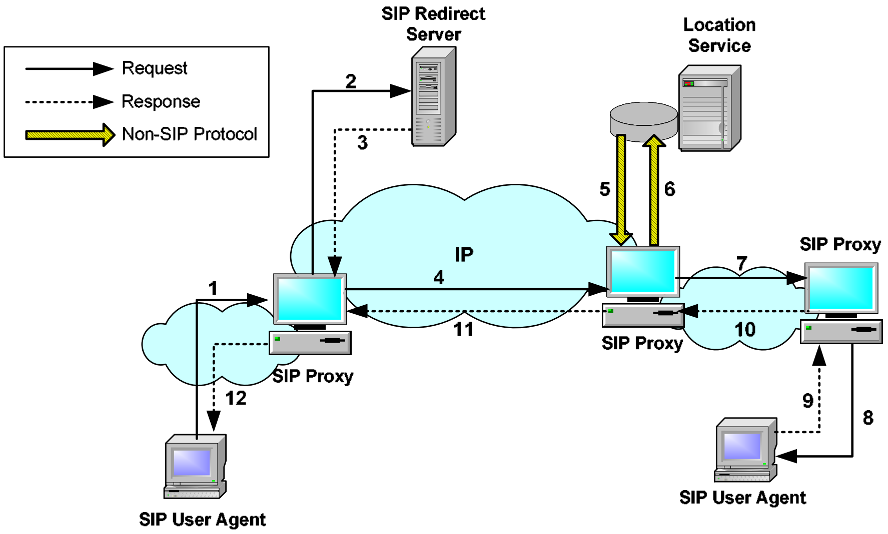

Figure 3: Interaction between SIP components

SIP methods

SIP is a lightweight protocol in that it has only a limited set of different message types (called in SIP terminology - methods). These methods, when combined together, allow for complete control over a multi-media call session while limiting complexity. There are two types of messages used in SIP:

- **Requests:** sent from the client to the server.

- **Responses:** sent from the server to the client.

Request methods:

INVITE	initiates a call, changes call parameters (re-INVITE)
ACK	confirms a final response for INVITE
BYE	terminates a call
CANCEL	cancels searching and "ringing"
OPTIONS	queries the capabilities of the other side
REGISTER	registers with the location service
INFO	sends mid-session information that does not modify the session state

Responses contain numeric response codes. The SIP response code set is partly based on HTTP response codes. There are two types of responses and six classes.

Response types:

PROVISIONAL (1xx class) - provisional responses are used by the server to indicate progress, but they do not terminate SIP transactions.

FINAL (2xx, 3xx, 4xx, 5xx, 6xx classes) - final responses terminate SIP transactions.

Classes:

2xx	provisional, searching, queuing, ringing, etc.
3xx	success
4xx	redirection, forwarding
5xx	server failures
6xx	global failure (busy, refusal, not available anywhere, etc.)

Addressing:

The "sip:" and "sips:" schemes use a form similar to the mailto URL, allowing the specification of SIP request-header fields and the SIP message-body. This makes it possible to specify the subject, media type, or urgency of sessions initiated by using a URI on a web page or in an email message.

H.323 and SIP in a protocol stack for the support of multimedia services

Multimedia protocol stack and the position of H.323 and SIP within it are depicted in figure 4.

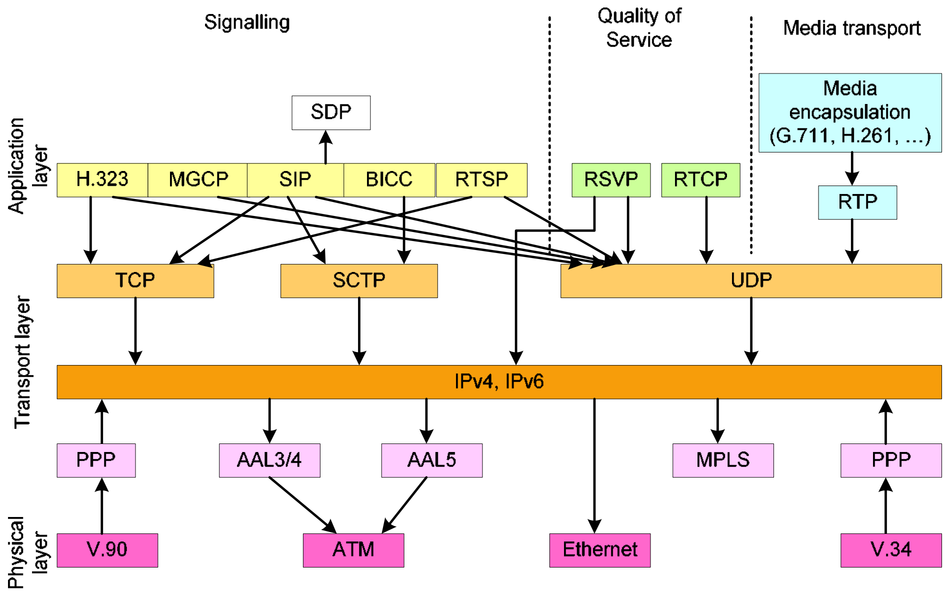

Figure 4: H.323 and SIP in a protocol stack for the support of multimedia services

5.4.2.2 Media Gateway control protocols

The necessity to interconnect the traditional telephone networks into NGN as well as the flexibility allowed by the separation of the NGN transport and control layers have lead to the distinction of the functions of the Media Gateway and Call/Session Server. This was the reason for developing a protocol which allows the Call/Session server to control the MG. In fact, it is a protocol which allows for the co-ordination between the transport and control layers. So far, two protocols for controlling the MG have been proposed: MGCP (Media Gateway Protocol) and MEGACO/H.248. These are relatively low-level device-control protocols that instruct an MG to connect streams coming from outside a packet network onto a packet stream such as the RTP (Real-Time Transport Protocol). Both the protocols follow the master/slave approach based on the distribution of the network gateway functions into "more intelligent" (master) "and less-intelligent" (slave) parts. This approach enables centralization of application intelligence in relatively fewer control servers (MGCs in the MGCP and MEGACO/H.248 terminology) on one hand and highly-cost and performance optimized gateway devices (MGs) on the other hand.

5.4.2.2.1 MGCP (Media Gateway Control Protocol)

MGCP (RFC 2705) was originally a proposal for the MEGACO/H.248. While MGCP had early deployment and is a reality in some networks, it is not representative of the current industry direction, nor is it a truly open standard. MGCP offers limited support of networks other than PSTN, is less flexible and extensible than MEGACO/H.248.

5.4.2.2.2 MEGACO/H.248

MEGACO (RFC 3015) defines the protocols used between elements of a physically decomposed multi-media gateway consisting of a MG and a MGC. MEGACO does not define how multiple MGs or MGCs communicate with each other. The standard is the result of a collaborative effort between the IETF WG MEGACO and ITU SG 16. Derived from MGCP (which, in turn, was derived from the combination of SGCP - Single Gateway Control Protocol and IPDC - IP Device Control Protocol), MEGACO draws heavily from MGCP plus introduces several enhancements. Even though MGCP was deployed first, MEGACO/H.248 is expected to be accepted as the official standard for decomposed gateway architectures sanctioned by both the IETF and ITU. MEGACO offers these key enhancements as compared to MGCP:

- Applicable to all packet network types, same service design for both IP and ATM.
- Supports multimedia and multipoint conferencing enhanced services.
- Provides for TCP and UDP transport options.
- Allows either text or binary encoding.

- Easy definition of new application interfaces through fully open package definition mechanism and IANA registration process.
- New packages can be defined based on existing packages.
- Allows package extension without affecting base protocol standard.

5.4.2.3 Signalling protocols between Call/Session Servers

There are two types of signalling protocols that might be used between Call/Session Servers:

- at the level of a core network (BICC, SIP-T, H.323);
- for the interconnection with the existing PSTN/ISDN networks, via the transport of PSTN/ISDN signalling over an IP-based network (SIGTRAN).

5.4.2.3.1 BICC (Bearer Independent Call Control)

BICC is a standard developed in ITU-T [12] and ETSI [13] for signalling. It has been proposed as the adaptation of the narrowband ISDN User Part (ISUP) for the support of narrowband ISDN services independent of the bearer technology and signalling message transport technology used. In terms of its name and origin it provides a common standard for signalling between networks that use different protocols, but in practice the design is strongly biased towards implementation on ATM networks and it suitability for pure IP networks is doubtful according to some experts. ITU-T Recommendation Q.1901 [12] describes the protocol for BICC CS1 (Capability Set). ITU-T Recommendation Q.1901 [12] is written as a delta document to ITU-T Recommendation Q.76x series. That is only the new formats, codes and procedures specific to BICC are described, additional to the ISUP Q.76x documentation. The Q.1902.x series describes BICC CS2. It supersedes the references to the ITU-T Recommendation Q.76X series as applied in ITU-T Recommendation Q.1901 [12] for BICC CS1.

5.4.2.3.2 H.323 between Call/Session Servers

In the IP network

In IP networks, ITU-T Recommendation H.323 use both Recommendations H.225 and H.245 for managing call control. Initially, these signalling channels were created between H.323 terminal (i.e. telephone set or H.323/ISDN gateway) and the H.323 call server. With the evolution of Recommendation H.323 which allows communication between H.323 call servers and which separates into distinct elements the MG and MGC functions, this Recommendation has been slightly modified.

It is stated now that call signalling and synchronization (H.225) go through between MGC; but the used protocol for exchanging capabilities between terminals, for channel negotiation and for media flow control between terminals H.323 (H.245) can be done between MG or MGC.

H.323 - SS7 interworking

H.323 interconnection with SS7 (ISUP): H.323 interconnection with the ISUP protocol has been specified recently in the Recommendation H.246 annex C. This annex describes the interworking between ISUP and H.225.0 Call Control protocol. It specifies the necessary mapping an Interworking Function would utilize to achieve connectivity and functionality between an H.323 network and an ISDN User Part network. The annex describes an interworking function when it is in a H.323 to PSTN gateway. The interworking function could reside in other elements of a H.323 network; this has been left for further study. The mapping described in this annex relates to a H.323 call to Circuit Switched Network Phone. The H.246 annex C shows how the ISUP services and functions would interwork with H.225.0. H.225.0 messages contain Q.931 information elements and as such parts of this annex have been derived from the Recommendation ITU-TQ.699. The annex does not show the mapping between H.320 and H.323.

H.246 annex E1 introduces the mechanisms to enable current mobile subscribers to access mobile networks using an H.323 terminal and obtain the same mobile services they get from a PLMN (Public Land Mobile Network) mobile terminal. The annex requires H.323 and H.225.0 versions 4 or later.

Tunnelling of ISUP in H.323: In order to support existing narrowband signalling information in an H.323 system, it is necessary to allow for transport of narrowband signalling information in H.323. Tunnelling of SS7 signalling protocols (ISUP) in H.323 has been specified in the Recommendation H.323 annex M2.

H.323 - DSS1 interworking: The guidance on tunnelling DSS1 over H.323 networks is given in the Recommendation H.323 annex E3. The annex requires H.323 and H.225.0 versions 4 or later.

H.323 - QSIG interworking: The guidance on how the generic tunnelling mechanism described in the Recommendation H.323 can be used to tunnel QSIG over H.323 networks is given in the H.323 annex M1.

5.4.2.3.3 SIP-T

SIP-T (SIP for telephones, previously SIP-BCP-T) [14] is a mechanism that uses SIP to facilitate the interconnection of the PSTN with IP-based networks. It is intended to allow traditional IN-type services to be seamlessly handled in the IP-based networks. It is essential that SS7 messages be available at the points of PSTN interconnection to ensure transparency of features not otherwise supported in SIP. SS7 messages should be available in their entirety and without any loss to the SIP network across the PSTN-IP interface. SIP-T defines SIP functions that map to ISUP interconnection requirements.

5.4.2.3.4 SIGTRAN

The SIGTRAN (Signalling Transport) protocol suite was developed within the IETF to allow an interworking between SS7 network elements and IP-based elements. Its primary purpose is to address the transport of PSTN/ISDN signalling over IP networks, taking into account the functional and performance requirements of the PSTN/ISDN signalling. To interwork with the PSTN/ISDN, IP-based networks need to transport signalling such as DSS1 (Digital Subscriber System), or SS7 (e.g. ISUP, SCCP) messages between IP nodes such a SG (Signalling Gateway), a MGC (Media Gateway Controller), and a (MG) Media Gateway.

The architecture that has been defined by SIGTRAN WG consists of three components (see figure 5):

- A standard IP.
- A common signalling transport protocol - a protocol that supports a common set of reliable transport functions for signalling transport. A SCTP (Stream Control Transmission Protocol) [15], [16] has been defined for this purpose by the IETF and endorsed by the ETSI (TS 102 144 [17]).
- An adaptation sub-layer that supports specific primitives, such as management indications, required by a particular signalling application protocol. The adaptation layer is different for different splits of the protocol. A number of different adaptation layers are being developed by the SIGTRAN group. Some of them have been recently endorsed by the ETSI TC SPAN.

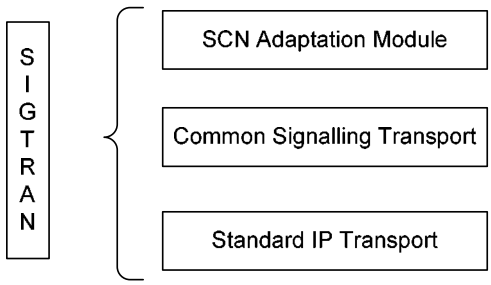

Figure 5: SIGTRAN architecture

- IUA: The boundary is Q.921/Q.931 [18].
- M2UA: The boundary is MTP2/MTP3 [19],[20].
- M2PA: The boundary is MTP2/MTP3, but for a symmetric scenario (SS7 MTP2-User Peer-to-Peer Adaptation Layer, Internet Draft - see bibliography).
- M3UA: The boundary is MTP3/user part [21],[22].
- SUA: The boundary is SCCP/SCCP user [23].

5.5 Service layer

In current telecommunication networks, the services are dedicated to the specific type of a network. An example might be the Intelligent Network over existing PSTN/ISDN dedicated mainly for the telephone terminals (fixed or mobiles), or services like mail and web dedicated for the IP-based networks.

Appearance of the new access technologies, like xDSL, cable networks, GPRS, UMTS, the tremendous growth of different types of terminal equipment, and the convergence of the core networks have lead to the necessity of the transformation of the architecture for the service platforms.

The principal question is how to make NGN profitable in order to enable a new class of services like: any-to-any ubiquitous communication services, IP multimedia services, and audio/video conferencing. All these services can be developed if NGNs address the main feature of service programmability. This means the ability of implementing new services following the customers' needs. Another important key point is the ability to offer to customers the same service everywhere, providing a seamless access from different terminals (mobile phones, soft-phones, UMTS phones, etc.). These goals can be obtained opening up levels of programmability to third parties or also to the users that want to personalize their own services.

There have been several models, or techniques defined for service development (Eurescom Project P1109: Service creation analysis in an NGN context - see bibliography).

5.5.1 OSA/Parlay

The current Intelligent Networks technology does not allow external service providers to create and deploy services on their own through the network of a network operator. The main reason is the missing security features in IN - a Service Creation Environment (SCE) has full access to the network operators signalling network SS7. Moreover, the third party service provider would have to invest a lot into the necessary equipment. To solve these issues, the Parlay group was founded in 1998 to specify and realize an open, technology-independent Application Programming Interface (API) in telecommunication networks. The Parlay API shall enable network operators, independent software manufacturers and service providers to offer products and services, which use the functionality of existing networks. This should not be restricted to one network type, but comprise various networks (see figure 6).

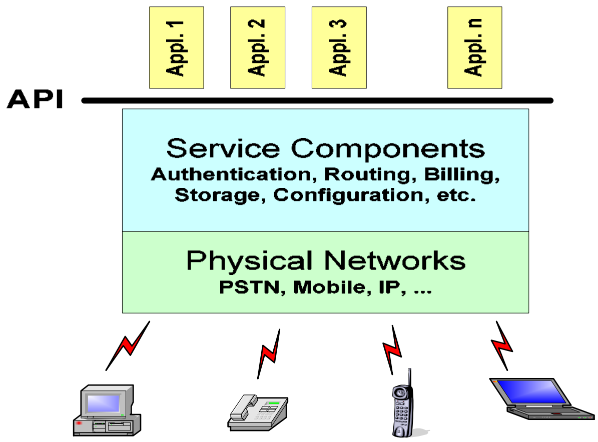

Figure 6: OSA architecture

5.5.1.1 Standardization of API

The efforts of the Parlay group to bring the API specification into standardization bodies succeeded already one year later. The 3GPP (3rd Generation Partnership Project), in charge of specifying 3G mobile networks, adapted Parlay as the method for creating services in UMTS. 3GPP introduced this API under the name OSA (Open Service Architecture) which recently was renamed to Open Service Access. The API thus is nowadays referred to as OSA/Parlay API (or vice versa). Meanwhile, the API has also been adapted by ETSI in order to cover the fixed network side. All three bodies jointly develop the standard. Although these bodies partly publish their own specifications, they are all aligned and compatible.

The Parlay group

Recently, the Parlay 4.0 specification (ES 202 915 [26]) has been defined jointly between ETSI, Parlay, and 3GPP, in cooperation with a number of Java APIs for Intelligent Networks (JAIN™) Community member companies. It is backward compatible with the recent Parlay 3.2 specifications (ES 201 915 [25]).

ETSI

ETSI publishes the master specification of the APIs. The Parlay Group point to this published ETSI specification.

3GPP maintain their own specification [24], which has the same structure as the ETSI specification. Each part of [24] is a subset or a copy of the ETSI specification.

- ETSI OSA Phase 1 [25] is identical to Parlay 3. 3GPP TS 29.198 Release 4 is a subset of this.
- ETSI OSA Phase 2 [26] is identical to Parlay 4. 3GPP TS 29.198 Release 5 is a subset of this.
- ETSI OSA Phase 3 (see bibliography) will be Parlay 5, and will correspond to 3GPP TS 29.198 Release 6.

5.5.1.2 OSA/Parlay architecture

The OSA/Parlay defines an architecture that enables the inter-working between the IT applications and the telecommunications features in the telecommunication network through an open standardized interface, i.e. the OSA/Parlay API's. The network functionality is described as SCFs (Service Capability Features) and applications could be deployed in a third party administrative domain. SCFs implement groups of Parlay/OSA APIs (e.g. Call Control APIs, Mobility APIs, Terminal Capabilities APIs, etc.) and provide access to the network capabilities that a Network Operator wants to export through OSA interface. They are provided/implemented by SCSs (Service Capability Servers) that are logical entities that implement one or more SCFs and interact with the network elements (e.g. SSP, HLR, Location server, etc.), as depicted in figure 7. The set of SCFs could be incrementally extended, since the OSA/Parlay's aim is to provide an extendible and scalable interface that allows for inclusion of new functionality in the network in future releases with a minimum impact on the applications using the OSA/Parlay interface.

The OSA/Parlay API consists of two groups of interfaces - "Framework Interfaces" and "Service Interfaces".

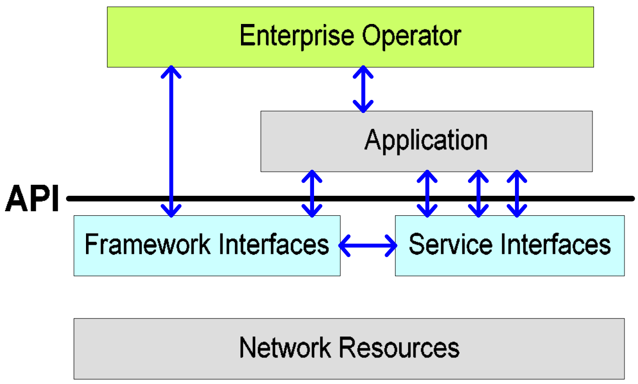

Figure 7: OSA/Parlay architecture

The **Framework Interfaces** provide basic mechanisms prior to the usage of actual network functions. They comprise, for instance, authentication and authorization to identify the application that wants to access the API. After successful authentication, the discovery function can be used to query information about availability of network functions. Further functions comprise online subscription of service features or network functions, and further contractual service usage agreements. The access to the Framework is always the first step for the use of the OSA/Parlay API. Following this, the Service Interfaces can be used, as far as the application is authorized.

The **Service Interfaces** enable client applications to access the so-called "Service Capability Features" (SCF). They represent the available network functions that can be used to implement telecommunication services for the end-customer. The following list gives an overview of the components contained in Parlay version 4.0, approved in 2002:

- Call Control: Setup and control of connections (Call Control Common Definitions, Generic Call Control, Multi-Party Call Control, Multi-Media Call Control).

- User Interaction: playing announcements, DTMF recognition, sending of SMS, etc.
- Mobility.
- Terminal Capabilities: to query to terminal capabilities.
- Data Session Control: e.g. for volume-based tariffying in GPRS.
- Generic Messaging: converting messages, connection to mailbox, etc.
- Connectivity Manager: realizing QoS, etc.
- Account Management: Management of prepaid cards in mobile networks.
- Charging.
- Policy Management.
- Presence and Availability Management.

OSA/Parlay APIs are provided as distributed APIs, through distributed processing mechanisms. Mapping on CORBA/IDL is already available, while mapping on Web Services technology (e.g. WSDL, SOAP) is under definition in Parlay X Working Group. Parlay Web Services Working Group has been created in order to define a model operational environment with client application, gateway, and application server roles, to identify requirements for implementation of Parlay environments using web services and to identify requirements for web services needed to support Parlay environments.

5.5.2 Web services

The main goal of Web services architecture is the realization of an interoperable network of services focused on service reuse and it is suitable both to interact with 3rd party applications and to export services by a network operator or a service provider. It is fully based on the Web-oriented architecture and protocols.

In September 2000, W3C (World Wide Web Consortium) started the XML (eXtensible Markup Language) Protocol Activity in order to address the need of an XML-based protocol for application-to-application messaging. In January 2002, the Web Services Activity was launched, subsuming the XML Protocol Activity by extending its scope to all the different aspects of Web services. The goal of the Web Services Activity is to design a set of technologies fitting in the Web architecture in order to bring the development of Web services to its full potential.

The Web Services can be used to export network services by exposing its WSDL (Web Services Definition Language) interfaces; these services communicate using SOAP (Simple Object Access Protocol), a protocol used to transport data between web services; service discovery and service registration are implemented accessing to the UDDI (Universal Discovery, Description and Integration) registry; XML is used as data format for SOAP messages that rely on existing internet protocols like HTTP. Web Services implementations need that the language-dependent API must be translated in WSDL and the application server where web-services are deployed must translate incoming SOAP messages to the underlying interfaces (Java, CORBA, etc.).

5.5.2.1 Protocols for "Web services"

5.5.2.1.1 SOAP (Simple Object Access Protocol)

SOAP is a XML-based object invocation protocol. It is a text-based protocol. SOAP was originally developed for distributed applications to communicate over HTTP and through corporate firewalls. SOAP defines the use of XML and HTTP to access services, objects and servers in a platform-independent manner.
SOAP does not itself define any application semantics such as a programming model or implementation specific semantics; rather it defines a simple mechanism for expressing application semantics by providing a modular packaging model and encoding mechanisms for encoding data within modules. SOAP consists of three parts:

- The SOAP envelope construct defines an overall framework for expressing what is in a message; who should deal with it, and whether it is optional or mandatory.
- The SOAP encoding rules defines a serialization mechanism that can be used to exchange instances of application-defined datatypes.

- The SOAP RPC representation defines a convention that can be used to represent remote procedure calls and responses.

In addition to the above, SOAP also defines two protocol bindings that describe how a SOAP message can be carried in HTTP messages either with or without the HTTP Extension Framework.

SOAP Version 1.2 includes two parts:

Part 1: "Messaging Framework" (W3C Candidate Recommendation, December 2002) (SOAP Version 1.2, Part 1 - see bibliography) defines, using XML technologies, an extensible messaging framework containing a message construct that can be exchanged over a variety of underlying protocols.

Part 2: "Adjuncts" (W3C Candidate Recommendation) (SOAP Version 1.2, Part 2 - see bibliography) defines a set of adjuncts that may be used with SOAP Version 1.2 Part 1.

5.5.2.1.2 WSDL (Web Services Description Language)

WSDL (specified by the W3C) is an XML format for describing network services as a set of endpoints operating on messages containing either document-oriented or procedure-oriented information. The operations and messages are described abstractly, and then bound to a concrete network protocol and message format to define an endpoint. Related concrete endpoints are combined into abstract endpoints (services). WSDL is extensible to allow description of endpoints and their messages regardless of what message formats or network protocols are used to communicate. However, the only bindings described in the WSDL Version 1.1 (specified in March 2001) describe how to use WSDL in conjunction with SOAP 1.1, HTTP GET/POST, and MIME. The WSDL Version 1.2 (Working Draft 3 published in March 2003 - see bibliography) defines a language for describing the abstract functionality of a service as well as a framework for describing the concrete details of a service description. The companion specification, WSDL Version 1.2: "Bindings" (see bibliography) defines a language for describing such concrete details for SOAP 1.2 and MIME.

5.5.2.1.3 UDDI (Universal Description, Discovery and Integration)

Technical matter related to Universal Description, Discovery and Integration (UDDI) is handled by UDDI.org's OASIS Technical Committees (TC). At the moment a single TC exists, the UDDI Spec TC (UDDI Specification TC). The UDDI specifications define a way to publish and discover information about Web services. The term "Web service" describes specific business functionality exposed by a company, usually through an Internet connection, for the purpose of providing a way for another company or software program to use the service.

For the process of Web service discovery, UDDI takes an approach that relies upon a distributed registry of businesses and their service descriptions implemented in a common XML format. The core component of the UDDI project is the UDDI business registration, an XML file used to describe a business entity and its Web services. Conceptually, the information provided in a UDDI business registration consists of three components: "white pages" including address, contact, and known identifiers; "yellow pages" including industrial categorizations based on standard taxonomies; and "green pages", the technical information about services that are exposed by the business. Green pages include references to specifications for Web services, as well as support for pointers to various file and URL based discovery mechanisms if required. UDDI includes the shared operation of a business registry on the Web. The UDDI Business Registry can be used at a business level to check whether a given partner has particular Web service interfaces, to find companies in a given industry with a given type of service, and to locate information about how a partner or intended partner has exposed a Web service in order to learn the technical details required to interact with that service. A UDDI registry exposes information about a business or other entity and its technical interfaces (or APIs). Two types of APIs are defined. A Subscription API defines the functions required for interactions between programs and the UDDI registry for the purpose of storing or changing data in the registry. A separate Subscription Listener API is provided for programs that want to access the registry to read the information from the registry. UDDI Version 3.0 (see bibliography) of the specification was published in July 2002. Expanding on the foundation of Versions 1 and 2, Version 3 offers the industry a specification for building flexible, interoperable XML Web services registries usable in private as well as public deployments.

5.5.3 Comparison of the two models

Both the models, OSA/Parlay and "Web services", have the same objective: to allows for a personalized access to multimedia services. For this, it is necessary to utilize a control layer that can control the interface between applications built by the use of the common service components and service resources: it is the API OSA/Parlay for the OSA model and a suite of SOAP, WSDL, and UDDI protocols for the "Web service" model.

The OSA/Parlay model (3GPP, Parlay, JAIN) is oriented towards a softswitch-based architecture, which being the central node of the control layer, presents the obligatory passage to access services (via OSA/Parlay interface). So, it seems to be adapted more to services which strongly depend on functions of a network control layer.

"Web services" model developed by the W3C, is based on "Web" technologies, so the architecture is distributed. Session initiation is done by the SIP. The adaptation layer, which is necessary for an access to services, is provided by the mix of "Web" protocols, like XML, SOAP, WSDL and UDDI. The "Web services" model is more oriented towards services relatively transparent for a network. It impacts mainly the service layer and partially the terminals, but it has only a little impact on a network itself.

5.5.4 Scripting languages

Scripting Languages are lightweight, highly customisable, and typically interpreted languages, appropriate in the area of rapid application development, acting as glue to provide connections among existing components. These characteristics allow them to be used to code or modify applications at runtime, and interact with running programs. These qualities and features make scripting languages applicable to the field of application programmability next to APIs.

Scripting languages represent, in an XML-based file, the service behaviour that can be changed at run-time; they act like a dynamic reconfiguration of the script interpreter that follows a pattern of registering the static events and criteria that can be matched by events by the underlying network components, followed by declaration of service logic that should be executed in response to such an event. Typically, scripts are created, edited, and validated using regular editors or as a result of applying transformation techniques.

The examples of scripting languages are: SCML (Service Creation Mark-up Language), CPL (Call Processing Language, Voice XML (Voice eXtensible Markup Language), CCXML (Call-Control eXtensible Markup Language).

5.5.5 SIP servlets

SIP servlets are a set of libraries that are used to create services on a SIP based network. The SIP Servlet API (SIP Servlet API Extensions - see bibliography) is a Java API based on the previously existing Servlet API. SIP Servlets are also a programming model where the Servlets (the applications) are hosted by an infrastructure known as a Servlet container. The SIP Servlet API allows application to initiate and to answer SIP requests. Therefore it simply exposes SIP capabilities (both User Agent and Proxy capabilities) to the application while hiding a few protocol details handled transparently by the SIP Servlet container. SIP Servlet API is suitable for third party service development. It could be noted that third party service development is rather simple since they are seen as Java libraries.

5.6 Issues affecting all layers

5.6.1 Addressing, numbering and naming

5.6.1.1 SCN (Switched Communication Networks)

One of the main advantages of the world-wide telephony network is the existence of a universal numbering scheme for subscribers that is agreed upon internationally under the auspices of the ITU [27]. This universal numbering plan allows the provision of the universal communication service: any subscriber of a sub-network telephony operator can be reached from any subscriber of any other sub-network operator in the world using the same E.164 [27] number allocated upon his subscription to his operator. This number is universal and, most important, its integrity and uniqueness is guaranteed world-wide. ITU-T Recommendation E.190 [28] describes the general principles to be utilized in the assignment of ITU-T Recommendation E-Series international numbering resources. ITU-T Recommendation E.164.1 [29] describes the procedures and criteria for the reservation, assignment, and reclamation of E.164 country codes and associated IC (Identification Code) assignments. The criteria and procedures are provided as a basis for the effective and efficient utilization of the available E.164 numbering resources.

5.6.1.2 IP-based networks

When IP was first standardized in September 1981, the specification required that each system attached to an IP-based Internet be assigned a unique, 32-bit Internet address value. Systems that have interfaces to more than one network require a unique IP address for each network interface. The first part of an Internet address identifies the network on which the host resides (Network Number, or Network Prefix), while the second part identifies the particular host on the given network (Host Number). This creates the two-level addressing hierarchy. To provide the flexibility required to support networks of varying sizes, the IP address space was be divided into three address classes-Class A, Class B, and Class C. This is often referred to as classful addressing. Each class fixes the boundary between the network prefix and the host number at a different point within the 32-bit address. In addition to the three most popular classes, there are two additional classes. Class D addresses have their leading four bits set to 1-1-1-0 and are used to support IP Multicasting. Class E addresses have their leading four bits set to 1-1-1-1 and are reserved for experimental use.

To make Internet addresses easier for people, IP addresses are often expressed as four decimal numbers, each separated by a dot ("dotted-decimal notation"). Dotted-decimal notation divides the 32-bit Internet address into four 8-bit fields and specifies the value of each field independently as a decimal number with the fields separated by dots.

In 1985, RFC 950 [30] defined a standard procedure to support the subnetting, or division, of a single Class A, B, or C network number into smaller pieces. Subnetting was introduced to overcome some of the problems that parts of the Internet were beginning to experience with the classful two-level addressing hierarchy. Instead of the classful two-level hierarchy, subnetting supports a three-level hierarchy. The basic idea of subnetting is to divide the standard classful host number field into two parts - the subnet number and the host number on that subnet.

In 1987, RFC 1009 specified how a subnetted network could use more than one subnet mask. When an IP network is assigned more than one subnet mask, it is considered a network with VLSM (Variable Length Subnet Masks) since the extended network prefixes have different lengths.

Throughout the Internet's growth, the problems concerning the near-term exhaustion of the Class B network address space and the rapid growth in the size of the global Internet's routing tables became critical and the response to these immediate challenges was the development of CIDR (Classless Inter-Domain Routing). CIDR was officially documented in September 1993 in RFC 1517 [31], 1518 [32], 1519 [33], and 1520 [34]. CIDR supports two important features that benefit the global Internet routing system. It eliminates the traditional concept of Class A, Class B, and Class C network addresses and it supports route aggregation where a single routing table entry can represent the address space of thousands of traditional classful routes. This allows a single routing table entry to specify how to route traffic to many individual network addresses.

With the growth of the Internet and its possible extension to additional devices, all IPv4 solutions proposed for scaling the Internet address space would only delay the inevitable, since there are just not enough IPv4 addresses. The IETF has produced a comprehensive set of specifications to define what is commonly known as the next-generation IP protocol ("IPng" or "IPv6"). IPv6 eliminates the need for VLSM, CIDR (Classless Inter-Domain Routing), and much more.

IPv6 increases the IP address size from 32 bits to 128 bits to support more levels of the addressing hierarchy, a much greater number of addressable nodes, and simpler auto-configuration. IPv6 supports approximately 340, 282, 366, 920, 938, 463, 374, 607, 431, 768, 211, 456 possible IP addresses. IPv6 text representation is very different from IPv4. The address form can be written three ways (preferred, compressed, and mixed) and it offers three different types of addresses (unicast, anycast, and multicast).

5.6.1.3 SCN-IP networks

One of the technical challenges raised by the integration between circuit-switched and IP networks is how to address calls that pass from one network service to another. Generally, it is assumed to be desirable to define an integrated global subscriber access plan. For example, the same ITU-T Recommendation E.164 [27] telephone number would reach a subscriber regardless of whether IP-based or PSTN network technologies are used.

ITU-T Recommendation SG 2 is currently studying a number of possible options whereby users in IP address-based networks can be accessed from/to SCN users. One option is the assignment of E.164 numbering resources to IP devices. Another approach is to support service interworking between different addressing systems in the SCN and IP networks; for example, using the IETF's ENUM protocol.

ENUM

ENUM is a protocol developed by the IETF's Telephone Number Mapping Working Group. The charter of the ENUM working group was to define a DNS-based (Domain Name System) architecture and protocol for mapping E.164 telephone numbers to internet domain names and linking them to communication services via what are known as URIs (Uniform Resource Identifiers). A relatively stable standardstrack version of the ENUM protocol has been published as RFC 2916 [35]. URIs are strings of characters that identify resources such as documents, images, files, databases, e-mail addresses or other resources or services in a common structured format. The most commonly known types of URIs are Uniform Resource Locators (URLs) which are used to locate resources using the World Wide Web. The syntax of URIs is defined in RFC 2396 [36].

The ENUM protocol uses so called NAPTR records (Naming Authority Pointer Resource Records) as defined in RFC 2915 in order to identify the available methods or services for contacting a specific node identified through an E.164 number. The ENUM protocol defines and uses a specific type of NAPTR record service with the mnemonic "E2U" (E.164 to URI resolution). The result of an ENUM query can be one or more URIs with their order of processing and preference indicated by values in the NAPTR records. These URIs are then used to reference resources or services associated with the E.164 number. Possible examples of resources or services include a fax number, mobile number, e-mail address, phone redirection services, unified messaging services, voice-mail and public key for asymmetric encryption applications.

In other words, ENUM allows anyone to know which communication channels are available in order to get in touch with someone else, using a simple telephone number, provided the called party has previously published his or her details (mobile phone number, fax number, IP telephony number, e-mail address, internet address, instant messaging address, etc.) in the ENUM "database".

At IETF level, RFC 2916 [35] is currently being revised. An initial draft revision of RFC 2916 [35] bis was published in July 2002. This draft is still under discussion.

Also, several drafts relating to ENUM have been published as part of the IETF's ENUM working group. Since IDs (Internet Drafts) are temporary documents which have a limited life, they are not listed in the present document.

ITU-T

RFC 2916 [35], describing the principles of ENUM, provides for delegating the management of zone "e164.arpa" in accordance with IAB instructions. It also suggests that the management of the domain names in this zone should be delegated in accordance with ITU-T recommendation E.164 [27], i.e. domain names in the zone "e164.arpa" corresponding to the country codes are delegated according to the same hierarchy as that laid down by recommendation E.164 [27] for country codes. To this end, the ITU-T, and more particularly Study Group 2, has been involved in defining the procedures between Tier-0 and Tier-1 entities (delegation of ENUM domain names corresponding to the country codes defined by recommendation E.164 [27]).

Some Member States of the ITU-T are opposed to using the domain name "e164.arpa" (under the control of IANA) and propose an independent domain name, under the control of ITU-T (or at least an international organization), to be used or set up if one does not already exist. The ITU-T Recommendation SG 2 has commissioned a group of experts to study the feasibility, advantages and disadvantages of establishing a new domain name specifically for ENUM.

In order to enable ENUM functionality tests to be carried out, the ITU-T Study Group 2 has drawn up interim procedures governing the administrative details between the RIPE-NCC (*Réseaux IP Européens* - Network Co-ordination Centre) and the ITU-T (more precisely, the Telecommunications Standardization Bureau, TSB) for delegating domain names in the zone "e164.arpa". These interim procedures will remain in force until the approval of one or more recommendations to replace them (currently: the draft E.A-ENUM recommendation). They can be amended at any time and will be re-examined later at the Study Group 2 meeting scheduled for May 2003.

Several documents have been drawn up or are in the process of being drawn up as part of the ITU-T's work on ENUM. Apart from the information document "Discussion on the global implementation of the ENUM protocol" published in September 2001, no documents about ENUM have been published by the ITU-T to date.

ETSI

On 12 July 2002, ETSI published a technical document [37] on ENUM administration in Europe based on the work of its SPAN11 NAR technical committee.

5.6.2 Media coding

This clause identifies the existing media coding technologies.

5.6.2.1 Voice coding

The purpose of voice coding is to transform a voice signal, generally analogue, into a digital signal of a given speed and quality.

5.6.2.1.1 Voice coding techniques in the context of the telephone network

Today's circuit switched telephone networks use, for the most part, coding systems based on the time technique, which is characterized by preservation of the waveform of the signal to be coded. According to the quantification method used, two types of coding can be identified: simple PCM (Pulse Coded Modulation) and differential coding (ITU-D IP Telephony Group of Experts Report - see bibliography).

PCM is the most simple of the coding algorithms used for coding speech in the PSTN and ISDN. This coding algorithm corresponds to ITU-T Recommendation G.711 [38] and to a data signalling rate of 64 kbit/s.

Differential coding (DPCM: Differential Pulse Code Modulation, ADPCM: Adaptive Differential Pulse Code Modulation, ADM: Adaptive Delta Modulation) is based on the observation that that successive samples from an audio source are highly correlated. It is therefore more advantageous to encode not the samples themselves, but the difference between the successive samples. ADPCM encoders encode the samples differentially with a component estimated by extrapolation from the preceding values. This coding algorithm, which corresponds to ITU-T Recommendation G.721, uses 32 kbit/s per voice channel.

The other two differential coding methods (DPCM and ADM) are characterized by the method used to predict the value of the following sample on the basis of the value of the preceding one. There are differential coding options which give data signalling rates of 16, 24 and 40 kbit/s; the speech quality deteriorates very rapidly when the rate falls to 16 kbit/s.

5.6.2.1.2 Voice coding techniques in the context of an IP network

The speech encoders currently used for voice coding in IP networks may be grouped according to three major coding techniques:

- Time techniques (bit rates between 16 and 64 kbit/s).
- Parametric techniques (bit rates between 2,4 and 4,8 kbit/s).
- Analysis-synthesis techniques (bit rates between 5 and 16 kbit/s).

The first category of encoders based on the time technique is widely used in conventional telephone networks and has been presented in the above paragraph. The latter two categories of coding techniques (parametric and synthesis) afford the advantage of low bit rates. On the other hand, the greater the compression rate, the longer the delay resulting from the processing stage will be. Thus, an optimum compromise needs to be always found between the bit rate and the associated processing delay.

The following table groups together, for the majority of coders mentioned above, the main characteristics in terms of bit rate, speech quality as an MOS (Mean Opinion Score), the average MOS being established in a standardized manner on the basis of five categories (1 = poor, 2 = mediocre, 3 = fairly good (average), 4 = good, 5 = excellent) and coding/decoding delay.

Table 6: Overview of selected voice coding standards

Coder	Standard	Bit rate	Quality of speech (MOS)	Coder/decoder delay
Time PCM	ITU-T Recommendation G.711 [38]	64 kbit/s	4,2	125µs
Time ADPCM	ITU-T Recommendation G.726 [39]	32 kbit/s	4,0	300µs
Analysis-synthesis RPE-LTP (Regular-Pulse Excitation with Long-Term Prediction)	ETSI I-ETS 300 036 [40] (GSM 06.10)	13 kbit/s	3,6	50 ms
Analysis-synthesis CELP (Code Excited Linear Predictive)	DOD FS1016	4,8 kbit/s	3,5	50 ms
Analysis-synthesis LD-CELP (Low-Delay - Code Excited Linear Predictive)	ITU-T Recommendation G.728 [41]	16 kbit/s	4,0	3 ms
Analysis-synthesis CS-ACELP (Conjugate Structure - Algebraic Code Excited Linear Prediction)	ITU-T Recommendation G.729 [42]	8 kbit/s	4,0	30 ms
Analysis-synthesis MP-MLQ-ACELP (MultiPulse - Maximum Likelihood Quantization)	ITU-T Recommendation G.723.1 [43]	6,3 kbit/s and 5,3 kbit/s	3,9 to 3,7	90 ms
Parametric LPC (Linear Predictive Coding)	DOD LPC10 FS1015	2,4 kbit/s	2,3	50 ms

5.6.2.1.3 Wideband speech coding techniques

Although most effort on speech coding focused on narrowband speech, the quality difference available by allowing the input speech to cover a larger bandwidth was recognized. The ITU-T established the first wideband speech coding standard, ITU-T Recommendation G.722, in 1998 [44]. ITU-T Recommendation G.722 [44] is simple to implement and achieves good performance at rates of 48 kbit/s, 56 kbit/s and 64 kbit/s. In 1995, ITU-T started another activity for wideband coding which resulted in 1999 in the standardization of ITU-T Recommendation G.722.1 at 24 and 32 kbit/s [45]. More recently, new wideband speech coding activities have been undertaken in ITU-T and ETSI 3GPP for coders at bit rates around 16 kbit/s. A new codec, known as AMR-WB (Adaptive Multi-Rate WideBand) codec [46] was selected in 2000 by 3GPP for wideband coding in GSM and 3G wireless systems. The AMR-WB codec participated in the ITU-T activity and was selected in 2001 as the winning candidate in the ongoing standardizing process in ITU-T for a wideband coder at rates of 13 kbit/s to 24 kbit/s.

Regional digital cellular standards have played a crucial role in the advancement of wireless second generation communication systems. For the GSM, several coders were standardized within the ETSI including the FR (full-rate) codec, the enhanced full-rate codec (EFR) and the AMR-WB codec mentioned above.

An overview of the selected wideband speech coding standards and their parameters is provided in table 7.

Table 7: Overview of selected wideband speech coding standards

Standard	ITU-T Recommendation G.722 [44]	ITU-T Recommendation G.722.1 [45]	3GPP/ETSI AMR-WB (ITU-T Recommendation G.722.2 [46])
Date	1988	1999	2000
Bit rate (kbit/s)	48, 56, 64 (embedded)	24, 32	23,85; 23,05; 19,85; 18,25; 15,85; 14,25; 12,65; 8,85; 6,6
Type	Sub-Band ADPCM	Transform Coding	ACELP (Algebraic Code Excited Linear Prediction)
Delay Frame size Lookahead	 0,125 ms 1,5 ms	 20 ms 20 ms	 20 ms 5 ms
Principle applications	ISDN, Video Conferencing	Same + VoPN (Voice over Packet Network)	3G wireless + Same as G.722.1 [45]

Currently, the work is going in the ITU-T Recommendation SG 16 on the voice coding algorithm specification G.4kbit/s. Its primary applications include very low-rate PSTN visual telephony, personal communications, simultaneous voice and data systems and mobile-telephony satellite systems.

The work in the ITU-SG 16 is going also on a new standard for variable rate coding of voice signals (G.VBR). Two technologies are being studied, MSC-VBR (Multi-rate Speech Coding - Variable Bit Rate) and Embedded ("EV"). Currently, terms of reference are being discussed in conjunction with the application areas for each of the studied technologies. Recommendations are expected in the 2003-2004 time frame.

IETF (iLBC) Internet Low Bit Rate Codec

iLBC (Internet Low Bit Rate Codec - see bibliography) is an algorithm for the coding of speech signals sampled at 8 kHz. iLBC uses a block-independent LPC (linear-predictive coding) algorithm and has support for two basic frame lengths: 20 ms at 15,2 kbit/s and 30 ms at 13,33 kbit/s. The two modes for the different frame sizes operate in a very similar way. The algorithm results in a speech coding system with a controlled response to packet losses similar to what is known from PCM with PLC (Packet Loss Concealment, such as the ITU-T Recommendation G.711 [38] standard which operates at a fixed bit rate of 64 kbit/s. At the same time, the algorithm enables fixed bit rate coding with a quality-versus-bit rate tradeoff close to state-of-the-art. A suitable RTP payload format for this codec is specified in the Internet draft document "RTP Payload Format for iLBC Speech" (RTP Payload Format for iLBC Speech - see bibliography).

Some of the applications for which this coder is suitable are: real time communications such as telephony and videoconferencing, streaming audio, archival, and messaging.

ETSI Aurora: Distributed Speech Recognition

Aurora is a working group within ETSI STQ TC (Speech processing, Transmission and Quality Aspects) that develops and standardizes an algorithm for DSR (Distributed Speech Recognition). The aim is to enable speech to be sent over low quality links such as mobile radio and converted to text for interacting with automated systems. The quality degradation of the links makes it necessary to perform a certain amount of pre-processing at the front end in the mobile terminal and send the results across the link for subsequent processing in the network. DSR will have widespread application in future mobile systems and will also be usable over Internet.

The primary objective of speech recognition is to enable to have easy access to the full range of computer services and communication systems, without the need to be able to type, or to be near a keyboard. By using a client/server approach in combination with the latest recognition systems, DSR will deliver the price/performance levels and access flexibility that will begin to make this practicable and affordable.

The performance of speech recognition systems receiving speech that has been transmitted over mobile channels can be significantly degraded when compared to using an unmodified signal. The degradations are as a result of both the low bit rate speech coding and channel transmission errors. A Distributed Speech Recognition (DSR) system overcomes these problems by eliminating the speech channel and instead using an error protected data channel to send a parameterized representation of the speech, which is suitable for recognition. The processing is distributed between the terminal and the network. The terminal performs the feature parameter extraction, or the front-end of the speech recognition system. These features are transmitted over a data channel to a remote "back-end" recognizer. The end result is that the transmission channel does not affect the recognition system performance and channel invariability is achieved.

Aurora is divided into two groups for the facilitation of work:

- AFE (Advanced Front End) group defining the front-end and speech processing related matters.
- A&P (Applications and Protocols) group that has been created to consider standards for Distributed Speech Recognition client-server protocols.

Two work items have been created by the AFE group. The first one proposes a front-end algorithm for DSR based on Mel-Cepstrum algorithm which performs in a low level of background noise environment. So far, ES 201 108 [47] presents this algorithm for a front-end to ensure compatibility between the terminal and the remote recognizer. The current target bit rate is 4,8 kbit/s but other rates like 9,6 kbit/s can be envisaged.

The second work item (DES/STQ-00008) will present another algorithm which should be able to match the Mel-Cepstrum algorithm performance in a more demanding environment like car, airport and so on, it aims to provide substantially improved performance in background noise. This has been measured in terms of reduction in error rate when evaluated on noisy speech databases covering a range of tasks and languages. The overall reduction in error rate is 53 % i.e. less than half the error rate when compared to the previous standard. This work will be made publicly available during 2002.

Two new work items, has recently been created to study the Front-end extension and Advanced Front-end extension for tonal language recognition and speech reconstruction. The purpose of this work item is to enable improved performance for tonal language recognition and to provide the ability to reconstruct a speech waveform form the DSR parameters.

5.6.2.1.4 Still-image coding

Classic facsimile (G3, G4) - ITU-T Recommendations T.4 [48]**, T.6 [**49**]**

The ITU-T Recommendation T.4 [48] defines the characteristics of Group 3 facsimile terminals which enable black and white documents and also optionally colour documents to be transmitted on the general switched telephone network, international leased circuits and the Integrated Services Digital Network (ISDN). Group 3 facsimile terminals may be operated manually or automatically and document transmission may be requested alternatively with telephone conversation. The procedures used by Group 3 facsimile terminals are defined in Recommendation T.30 [86].

The ITU-T Recommendation T.6 [49] defines the facsimile coding schemes and their control functions to be used in the Group 4 facsimile.

JBIG (Joint Bi-level Image experts Group) standards:

JBIG is a group of experts nominated by national standards bodies and major companies to produce standards for bi-level image coding. The "joint" refers to its status as a committee working on both ISO and ITU-T standards. The official title of the committee is ISO/IEC JTC1 SC29 Working Group 1, and is responsible for both JPEG and JBIG standards.

JBIG have developed **JBIG1** (ISO/IEC 11544 [50]/ITU-T Recommendation T.82 [51]) for the lossless compression of a bi-level image. It can also be used for coding greyscale and colour images with limited numbers of bits per pixel. It can be seen as a form of facsimile encoding, similar to Group 3 or Group 4 fax, offering between 20 and 80 % improvement in compression over these methods (about 20 to one over the original uncompressed digital bit map).

JBIG2 (ISO/IEC 14492) [52] defines a coding method for bi-level images (e.g. black and white printed matter). These are images consisting of a single rectangular bit plane, with each pixel taking on one of just two possible colours. Multiple colours are to be handled using an appropriate higher level standard such as ITU-T Recommendation T.44 [53]. Compression of this type of image is also addressed by existing facsimile standards, for example by the compression algorithms in ITU-T Recommendations T.4 [48], T.6 [49], T.82 [51] (JBIG1), and T.85 (Application profile of JBIG1 for facsimile) [54]. Besides the obvious facsimile application, JBIG2 is be useful for document storage and archiving, coding images on the World Wide Web, wireless data transmission, print spooling, and even teleconferencing.

Unlike JBIG 1, JBIG2 was explicitly prepared for lossy, lossless, and lossy-to-lossless image compression. The design goal for JBIG2 was to allow for lossless compression performance better than that of the existing standards, and to allow for lossy compression at much higher compression ratios than the lossless ratios of the existing standards, with almost no visible degradation of quality. In addition, JBIG2 allows both quality-progressive coding, with the progression going from lower to higher (or lossless) quality, and content-progressive coding, successively adding different types of image data (for example, first text, then halftones). A typical JBIG2 encoder decomposes the input bi-level image into several regions and codes each of the regions separately using a different coding method. Such content-based decomposition is very desirable especially in interactive multimedia applications. JBIG2 can also handle a set of images (multiple page document) in an explicit manner. As is typical with image compression standards, JBIG2 explicitly defines the requirements of a compliant bitstream, and thus defines decoder behaviour. JBIG2 does not explicitly define a standard encoder, but instead is flexible enough to allow sophisticated encoder design. In fact, encoder design is a major differentiator among competing JBIG2 implementations.

JPEG (Joint Photographic Expert Group)

JPEG1 is the ISO/ITU-T (ISO/IEC 10918-4:1999 [55]/ITU-T Recommendation T.870 [56]) standard. There are several modes defined for JPEG, including baseline, lossless, progressive and hierarchical. The baseline mode is the most popular one and supports lossy coding only.

In the baseline mode, the image is divided in 8x8 blocks and each of these is transformed with the DCT (Discrete Cosine Transformation). The transformed blocks are quantized with a uniform scalar quantizer, zig-zag scanned and entropy coded with Huffman coding. The quantization step size for each of the 64 DCT coefficients is specified in a quantization table, which remains the same for all blocks. The DC coefficients of all blocks are coded separately, using a predictive scheme. This is a mode simply referred as JPEG.

The lossless mode is based on a completely different algorithm, which uses a predictive scheme. The prediction is based on the nearest three causal neighbours and seven different predictors are defined (the same one is used for all samples). The prediction error is entropy coded with Huffmann coding. This mode is referred as L-JPEG.

The progressive and hierarchical modes of JPEG are both lossy and differ only in the way the DCT coefficients are coded or computed, respectively, when compared to the baseline mode. They allow a reconstruction of a lower quality or lower resolution version of the image, respectively, by partial decoding of the compressed bitstream.

MPEG-4 VTC (Moving Pictures Experts Group - Visual Texture Coding):

MPEG-4 VTC is the algorithm used in MPEG-4 to compress visual textures and still images, which are then used in photo realistic 3D models, animated meshes, etc., or as simple still images. It is based on the DWT (Discrete Wavelet Transform), scalar quantization, zero-tree coding and arithmetic coding.

A unique feature of MPEG-4 VTC is the capability to code arbitrarily shaped objects. This is accomplished by the means of a shape adaptive DWT and MPEG-4's shape coding. Several objects can be encoded separately, possibly at different qualities, and then composed at the decoder to obtain the final decoded image. On the other hand, MPEG-4 VTC does not support lossless coding.

JPEG-LS (Lossless)/ITU-T Recommendation T.86 [57**]:**

JPEG-LS is the ISO/ITU-T standard [57] for lossless coding of still images. It also provides for "near-lossless" compression. Part-I, the baseline system, is based on adaptive prediction, context modelling and Golomb coding. In addition, it features a flat region detector to encode these run-lengths. Near-lossless compression is achieved by allowing a fixed maximum sample error. Part-II introduces extensions such as an arithmetic coder. This algorithm was designed for low-complexity while providing high lossless compression ratios. However, it does not provide support for scalability, error resilience or any such functionality.

PNG (Portable Network Graphic):

PNG is a W3C recommendation for coding of still images which has been elaborated as a patent free replacement for GIF, while incorporating more features. It is based on a predictive scheme and entropy coding. The prediction is done on the three nearest causal neighbours and there are five predictors that can be selected on a line-by-line basis. PNG is capable of lossless compression only and supports grey scale, palet colour and true colour, an optional alpha plane, interlacing and other features.

JPEG 2000/ITU-T Recommendation T.800 [59**]:**

JPEG 2000 is the new ISO [58]/ITU-T [59] standard for still image coding. The Part 1 of the standard defines the core system. JPEG 2000 is based on the DWT, scalar quantization, context modelling, arithmetic coding and post-compression rate allocation. One of the most interesting features is that JPEG 2000 can accommodate lossless compression as well as lossy compression in the same file, the lossy one being part of the total file. Simply cutting short transmission allows an application to provide a reduced quality or lower resolution image directly from a lossless archive. This can be seen as a possible replacement for previous data structures where a lossy image (an original JPEG image for instance) was kept separately from the reference lossless content, possibly stored as TIFF (Tagged Image File Format), PNG or other reference format. It is particularly relevant in application fields where lossless is needed for downstream processing and archiving, whilst lossy transmission is required for fast viewing (medical, satellite imaging, remote sensing applications, for instance). A simple file format is also included (JP2 files).

Part 2 **"Extensions"** provides various extensions for specific applications.

Part 3 **"Motion JPEG 2000"** (ITU-T Recommendation T.802) allows the same algorithms to be used on a time-based frame by frame basis, allowing moving and still sequences to be treated as closely aligned., whilst preserving such baseline capabilities as lossless image coding if needed.

Part 4 **"Compliance testing"** provides test sequences which exercise the complexity of the standard by identifying correct coding behaviour.

Part 5 **"Reference software"** aims at making available to the public some reference software, a process that was one of the best boosters for the current JPEG.

Part 6 **"Compound image file format"** defines a syntax to describe images where different types of content are put into a compound file, to optimize the code and take advantage of other compression algorithms. Parts of the imaged document can use JPEG's sister committees, JBIG, algorithms or even existing code such as facsimile or original JPEG images.

Part 7 was initially about hardware implementation but has now been withdrawn.

Part 8 **"JPSEC Secure JPEG 2000"** is one of the four new work items initiated in 2001, addressing security in terms of content, considering methods to encrypt, scramble, watermark contents to help establish trust in the future e-commerce marketplace. Conditional access, registration, identification and traffic monitoring are considered in this work item.

Part 9 **"JPIP interactivity tools APIs and protocols"** deals with interactivity, especially over Internet, delivery of metadata and the use of protocols to perform functions such as displaying different levels of quality or resolution according to the user's needs and available bandwidth.

Part 10 **"JP3D 3-D and floating point data"** is addressing 3D and volumetric imaging, including floating point data, allowing for extension of the image concept in medical and volumetric representations.

Part 11 **"JPWL wireless"** is about wireless image transmission, for instance when an image projector is "connected" wirelessly to a laptop, but primarily to help robust transmission of images in the noisy environment of cell-phone links.

Part 12 **"ISO base media file format"** is a part that has been introduced recently as the common work item with MPEG where a compliant file format will be used for both MPEG-4 and Motion JPEG 2000 images, to provide a richer environment for users of such technology.

Parts 1 to 5 have reached the level of the International Standard, Part 6 is in the FDS (Final Draft Status), Part 8 to 11 are at various stages in the Working Draft process and work on Part 12 has just started. The JPEG 2000 has many strong points and offers the additional functionalities for applications ranging from secure archives (lossless encoding) to Motion JPEG 2000 for digital cinema (no restrictions in dimensions of images in multiple formats using part of the code stream), and medical images. The attention to detail in metadata handling and its integration with the physical image allows for content protection and access control by introducing security aspects within the file.

5.6.2.1.5 Moving image coding

ITU-T Recommendation H.120 [60**]**

ITU-T Recommendation H.120 [60] specifies codecs for videoconferencing using primary digital group transmission. It specifies how 625-line and 525-line television camera signals can be converted to a digital format suitable for transmission over broadband telephone services. ITU-T Recommendation H.120 [60] can be used for transmission of data between television service providers, but it can be also used for the transmission of camera-captured information as part of a videoconferencing link using broadband lines.

ITU-T Recommendation H.261 [61**]**

ITU-T Recommendation H.261 [61] describes the video coding and decoding methods for the moving picture component of audiovisual services at the rates of $p \times 64$ kbit/s, where p is in the range 1 to 30.

H.262 [62**] (same as ISO/IEC 1318-2 MPEG-2/Video)**

H.263 (H.263+, H.263++)

ITU-T Recommendation H.263 Version 1 specifies a coded representation that can be used for compressing the moving picture component of audio-visual services at low bit rates. The basic configuration of the video source coding algorithm is based on Recommendation H.261 [61] and is a hybrid of inter-picture prediction to utilize temporal redundancy and transform coding of the remaining signal to reduce spatial redundancy. The source coder can operate on five standardized picture formats: sub-QCIF, QCIF, CIF, 4CIF and 16CIF.The decoder has motion compensation capability, allowing optional incorporation of this technique in the coder. Half pixel precision is used for the motion compensation, as opposed to Recommendation H.261 [61] where full pixel precision and a loopfilter are used. Variable length coding is used for the symbols to be transmitted.

In addition to the basic video source coding algorithm, four negotiable coding options are included for improved performance: Unrestricted Motion Vectors, Syntax-based Arithmetic Coding, Advanced Prediction and PB-frames. All these options can be used together or separately. H.263 has been adopted in several videophone terminal standards, notably ITU-T Recommendation H.324 (PSTN), H.320 (ISDN), and H.310 (B-ISDN), 3GPP.

H.263 Version 2, also known as "H.263+", was officially approved in September 1997. H.263+ is an extension of H.263, providing 12 new negotiable modes and additional features. These modes and features improve compression performance, allow the use of scalable bit streams, enhance performance over packet-switched networks, support custom picture size and clock frequency, and provide supplemental display and external usage capabilities.

H.263 Version 3, also known as "H.263++" defines new normative "profiles" and "levels" for H.263. Annex X of the Recommendation contains normative profile and level definitions.

MPEG (Moving Pictures Expert Group) and its standards:

MPEG (formally ISO/IEC JTC1/SC29/WG11) is a group of experts dedicated to the development of compression and decompression standards for digitizing and delivering audio, video and multimedia over computers systems and networks, including the web.

MPEG-1 (ISO/IEC 11172-2 [63]) standard was approved in November 1992 as a complete audio-visual solution for interactive video and audio broadcasting. MPEG-1 is intended to be generic in a sense that the standard is independent of a particular application and therefore comprises mainly a toolbox. It is up to the user to decide, which tools to select to suit the particular applications envisaged. The bit rates involved were about 1,5 Mbit/s for audio and video together and 256 kbit/s for audio only.

MPEG-2 (ISO/IEC 13818 [64]/ITU-T. H.262 [62]) was developed in 1994 as a complete digital audio-video solution for broadcasting and interactive video. The video coding scheme used in MPEG-2 is again generic and similar to the one of MPEG-1, however with further refinements, such as "scalability", and special consideration of interlaced sources.

The bit rate below 1 Mbit/s was not specifically addressed by MPEG-1 and MPEG-2. This is a reason why high-quality video coding at bit rates as low as 10 kbit/s was the goal of the MPEG-4 Video project. Version 1 of MPEG-4 was approved in October 1998 and version 2 in December 1999.

MPEG-4 (ISO/IEC 14496-2 [65]) also known as MPEG4-Video, was developed in response to the growing need for a coding method that can facilitate access to visual objects in natural and synthetic moving pictures and associated natural or synthetic sound for various applications such as digital storage media, internet, various forms of wired or wireless communication. This means that motion video can be manipulated as a form of computer data and can be stored on various storage media, transmitted and received over existing and future networks and distributed on existing and future broadcast channels.

In particular, MPEG-4 addresses for:

Universal accessibility and robustness in error-prone environments: Multimedia audio-visual data need to be transmitted and accessed in heterogeneous network environments, possibly under severe error conditions (e.g. mobile channels). Although the MPEG-4 standards is network (physical-layer) independent in nature, the algorithms and tools for coding audio-visual data have been designed with awareness of network peculiarities.

High interactive functionality: Multimedia applications call for extended interactive functionalities to assist the user's needs. In particular the flexible, highly interactive access to and manipulation of audio-visual data are of prime importance. It is envisioned that - in addition to conventional playback of audio and video sequences - the user needs to access "content" of audio-visual data to present and manipulate/store the data in a highly flexible way.

Coding of natural and synthetic data: Next generation processors will enable multimedia terminals to present sample-based audio and pixel-based video together with synthetic audio/speech and video in a highly flexible way. MPEG-4 assists the efficient and flexible coding and representation of both natural (sample-based) as well as synthetic data. Compression efficiency: For the storage and transmission of audio-visual data, a high coding efficiency, meaning a good quality of the reconstructed data, is required. Improved coding efficiency, in particular at very low bit rates below 64 kbits/s, continues to be an important functionality that is supported by the MPEG-4 video standard.

The applications of MPEG-4 cover, but are not limited to, internet multimedia, interactive video games, interpersonal communications such as videoconferencing, interactive storage media, multimedia mailing, networked database services, remote emergency systems, remote video surveillance, wireless multimedia and multimedia in general.

MPEG-4 Video is being used in a number of environments. The most promising is for mobile communication applicable to the GPRS and to the UMTS. The next is to further compress movie files (themselves compressed using MPEG-2) so that they can fir into a single CD-ROM. Another one is for video streaming over Internet.

MPEG-4 version 1 was approved in 1998, version 2 in 1999. MPEG is currently working on MPEG-4 versions 3, 4 and 5.

H.264/MPEG-4 AVC

The H.264/MPEG-4 AVC video compression standard promises a significant improvement over all previous video compression standards. In terms of coding efficiency, the new standard is expected to provide at least 2x compression improvement over the best previous standards and substantial perceptual quality improvements over both MPEG-2 and MPEG-4. The standard, being jointly developed by ITU-T and ISO/IEC, addresses the full range of video applications including low bit rate wireless applications, standard-definition and high-definition broadcast television, video streaming over the Internet, delivery of high-definition DVD content, and the highest quality video for digital cinema applications.

The ITU-T name for the standard is H.264 (previously called H.26L), while the ISO/IEC name is MPEG-4 Advanced Video Coding (AVC) which will become Part 10 of the MPEG-4 standard. Since AVC is an extension to the current MPEG-4 standard, it benefits from MPEG-4's well-developed infrastructure tools (e.g. system layer and audio). It is expected that MPEG-4 AVC will be selected over the current MPEG-4 video compression standard, known as MPEG-4 Advanced Simple Profile (ASP), for the majority of applications that demand the highest compression and quality levels. It is expected that H.264/MPEG-4 AVC will displace MPEG-2 and MPEG-4 ASP in many existing applications, in addition to opening up several new markets (e.g. video over ADSL).

MPEG-7

MPEG-7, formally named "Multimedia Content Description Interface", is a standard [66] for describing the multimedia content data that supports some degree of interpretation of the information's meaning, which can be passed onto, or accessed by, a device or a computer code. MPEG-7 is not aimed at any one application in particular; rather, the elements that MPEG-7 standardizes support as broad a range of applications as possible. Audiovisual data content that has MPEG-7 data associated with it, may include: still pictures, graphics, 3D models, audio, speech, video, and composition information about how these elements are combined in a multimedia presentation (scenarios). A special case of these general data types is facial characteristics. Thus, MPEG-7 addresses applications that can be stored (on-line or off-line) or streamed (e.g. broadcast, push models on the Internet), and can operate in both real-time and non real-time environments. A "real-time environment" in this context means that the description is generated while the content is being captured. MPEG-7 Description Tools do, however, not depend on the ways the described content is coded or stored.

MPEG-7 data may be physically located with the associated AV material, in the same data stream or on the same storage system, but the descriptions could also live somewhere else on the globe. When the content and its descriptions are not co-located, mechanisms that link the multimedia material and their MPEG-7 descriptions are needed; these links will have to work in both directions.

MPEG-7 addresses many different applications in many different environments, which means that it needs to provide a flexible and extensible framework for describing audiovisual data. Therefore, MPEG-7 does not define a monolithic system for content description but rather a set of methods and tools for the different viewpoints of the description of audiovisual content. Having this in mind, MPEG-7 is designed to take into account all the viewpoints under consideration by other leading. These standardization activities are focused to more specific applications or application domains, whilst MPEG-7 has been developed as generic as possible. MPEG-7 uses also XML as the language of choice for the textual representation of content description, as XML Schema has been the base for the DDL (Description Definition Language) that is used for the syntactic definition of MPEG-7 Description Tools and for allowing extensibility of Description Tools (either new MPEG-7 ones or application specific). Considering the popularity of XML, usage of it will facilitate interoperability with other metadata standards in the future.

The main elements of the MPEG-7's standard are:

- Description Tools: Descriptors (D), that define the syntax and the semantics of each feature (metadata element); and Description Schemes (DS), that specify the structure and semantics of the relationships between their components, that may be both Descriptors and Description Schemes.

- A Description Definition Language (DDL) to define the syntax of the MPEG-7 Description Tools and to allow the creation of new Description Schemes and, possibly, Descriptors and to allow the extension and modification of existing Description Schemes.

- System tools, to support binary coded representation for efficient storage and transmission, transmission mechanisms (both for textual and binary formats), multiplexing of descriptions, synchronization of descriptions with content, management and protection of intellectual property in MPEG-7 descriptions, etc.

MPEG-21

Work on the new standard MPEG-21 started in June 2001. MPEG-21 aims at defining a normative open framework for multimedia delivery and consumption for use by all the players in the delivery and consumption chain. This open framework will provide content creators, producers, distributors and service providers with equal opportunities in the MPEG-21 enabled open market. This will also be to the benefit of the content consumer providing them access to a large variety of content in an interoperable manner.

MPEG-21 is based on two essential concepts: the definition of a fundamental unit of distribution and transaction (the Digital Item) and the concept of Users interacting with Digital Items. The Digital Items can be considered the "what" of the Multimedia Framework (e.g. a video collection, a music album) and the Users can be considered the "who" of the Multimedia Framework.

The goal of MPEG-21 is thus to define the technology needed to support Users to exchange, access, consume, trade and otherwise manipulate Digital Items in an efficient, transparent and interoperable way. MPEG-21 identifies and defines the mechanisms and elements needed to support the multimedia delivery chain as well as the relationships between and the operations supported by them. Within the parts of MPEG-21, these elements are elaborated by defining the syntax and semantics of their characteristics, such as interfaces to the elements.

VRML (Virtual Reality Modelling Language)

Originally a proprietary standard the specification was placed in the public domain and formally adopted by the VRML Architecture Group (VAG). During 1999 the Web 3D consortium took over promotion of the standard. The specification has been standardized by ISO/IEC JTC1/SC24.

A subset of Silicon Graphic's Inventor File Format, with extensions to support interworking, formed the basis of Version 1.0 of the VRML that is used for describing multi-participant interactive simulations - virtual words - networked via the Internet and hyperlinked with the WWW.

VRML97, which has been standardized as IS 14772-1, allows for richer behaviours, including animations, motion physics and real-time multi-user interaction. Part 2 of the standard defines EAI (External Authoring Interface).

X3D (Extensible 3D)

An X3D application specified within the Web 3D consortium Extensible 3D Graphics Working Group is a 3D time-based space that contains graphic and aural objects that can to enable dynamically modified. Part 1 of the specification provides an abstract functional specification for the X3D framework, and definitions of the standardized components and profiles. Part 2 contains the abstract functional specification for APIs to the X3D runtime system, and bindings to various programming languages and component object models. Part 3 contains the data encoding specification for an XML (eXtensible Markup Language) encoding of X3D, and Part 4 contains the data encoding specification for the VRML97 utf-8 encoding of X3D.

Five restricted profiles, in addition to the full X3D profile will be defined in the standard to enable compatibility with existing applications of virtual reality browsers: interchange profile, interactive profile, MPEG-4 interactive profile, extensibility profile and VRML97 profile.

Final working draft published in July 2002 was submitted to ISO as a first Committee draft for a standard.

5.6.2.1.6 ITU-R activities in broadcasting

ITU-R BS. 1115 (Low bit-rate audio coding) [67]

ITU-R Recommendation BS. 1115 [67] addresses two-channel low bit-rate audio coding to be used for digital sound broadcasting applications. This recommendation specifies the audio coding systems and operational bit rate for four broadcasting applications: contribution, distribution, emission, and commentary. For emission, this recommendation prescribes MPEG-1 Layer II at a bit rate of 128 kbit/s per channel. For contribution and distribution, it specifies MPEG-1 Layer II at a bit rate of at least 180 kbit/s per channel, and for commentary, MPEG-1 Layer III at a bit rate of at least 60 kbit/s per channel.

ITU-R BS. 1196-1 (Audio coding for DTTB) [68**]**

MPEG has standardized MPEG-2 BC (Backward Compatible) and MPEG-2 AAC (Advanced Audio Coding) coding for multi-channel audio. ITU-R Recommendation BS.1196 has specified MPEG-2 BC and AC-3 non-backward compatible compression and coding as audio coding systems for DTTB (Digital Terrestrial Television Broadcasting). In Europe, MPEG-2 BC is used for DVB system and DVD.

ITU-R BT. 1208 (Video coding for DTTB) [69**]**

The MPEG-2 video-encoding profile/level combinations available as "conformance points" have been reduced to two within this Recommendation. These two profiles/levels are:

- Main Profile at Main Level (MP@ML) which defines SDTV (Standard-definition Digital Television) services.
- Main Profile @ High Level (MP@HL) which provides for both HDTV (High-definition Television) services and SDTV services.

The set of DTTB video subsystem tools are defined in ITU-R Recommendation BT.1208 [69] and allow content producers to provide programming in conventional, widescreen, and HDTV formats.

ITU-R Recommendation SG 6

There has been a new question "Digital cinema (D-cinema) broadcasting" created in the frame of ITU-R Recommendation SG 6 TG6/9 that includes a number of questions relating not only to coding, but also to performance, assessment, digital production, recording formats, operation practices, and delivery of D-cinema programs.

Multimedia Broadcasting

Multimedia broadcasting may be the most important frontier of technology in digital broadcasting today. The most important technical issue to resolve is the language that is used for multimedia applications that are broadcast over-air. This is the API (Application Programming Interface). In this respect, the ITU-R has already produced an ITU-R Recommendation BT.1378: "Basic requirements for multimedia-hypermedia broadcasting" [70]. It recommends initial basis for multimedia-hypermedia broadcasting requirements. This matter was considered so critical for broadcasting, that the ITU-R proposed setting up a new group to work actively in this area over the coming months to agree a common worldwide system (ITU-R Recommendation WP 6M "Interactivity and multimedia broadcasting").

5.6.3 QoS and performance

5.6.3.1 Definitions and framework

There are several candidate definitions of quality and QoS (Quality of Service). While ISO 8402 [71] provides a general definition of quality itself, ITU-T Recommendation E.800 [72] provides a definition of QoS.

A definition of general quality is provided in ISO 8402 [71] as "the totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs".

Similarly, ISO 9000 [73] defines quality as the "degree to which a set of inherent characteristics fulfils requirements". The ISO 8402 [71] definition seems better from the user's view. In the both cases event, QoS is clearly a subset of overall quality.

ITU-T Recommendation E.800 [72] defines QoS as "the collective effect of service performance which determine the degree of satisfaction of a user of the service".

Criteria for judging the quality of the communication functions that any service must support are given in [74]. However, even those criteria can be viewed from different perspectives:

Customer's QoS requirements: They state the level of quality required of a particular service, which may be expressed in non-technical language. The customer is not concerned with how a particular service is provided, or with any aspects of the network's internal design, but only with the resulting end-to-end service quality.

Service provider's offerings of QoS (targeted QoS): This is a statement of the level of quality expected to be offered to the customer by the service provider. The level of quality is expressed by values assigned to QoS parameters. The principal use of this form of QoS is for planning and for SLAs (Service Level Agreements). Each service would have its own set of QoS parameters. The service provider may express the offered QoS in non-technical terms for the benefit of customers, and in technical terms for use within the business.

QoS achieved/delivered: This is a statement of the level of quality actually achieved and delivered to the customer. It is expressed by values assigned to parameters, which should be the same as specified for the offered QoS so that the two can be compared to determine what was actually achieved to assess the level of performance achieved. These performance figures are summarized for specified periods of time, e.g. for the previous month.

Customer survey ratings of QoS: This is a statement expressing the level of quality experienced they "believe" they have experienced. The Perceived QoS is expressed, usually in terms of degrees of satisfaction and not in technical terms. Perceived QoS is assessed by customer surveys and from customer's own comments on levels of service. Perceived QoS can be used by the service provider to determine customer satisfaction of the service quality.

5.6.3.2 End-to-end QoS for multimedia services and applications

It has to be taken into account that multimedia services and applications will have to support different kinds of audio, video and data (e.g. text, still images, graphics) combinations. However, each medium will have its own QoS requirements that are expressed to satisfy the needs of the user. These requirements are expressed on the end-to-end basis between the users and are independent of the underlying transport network. The QoS requirements can be divided into two groups:

- QoS requirements related to the quality of media transmission (i.e. the users need to negotiate QoS parameters on end-to-end basis to satisfy the requirements of the application that might consist of audio codecs, video codecs, and/or data applications).
- QoS requirements related to quality of signalling (i.e. quality of control mechanisms that are invoked by the application between the communication ends before sending any media and that provide a mechanism to obtain information including the QoS parameters how the media to be sent and received between the communicating ends).

5.6.3.3 End-user multimedia QoS categories

A typical user is not concerned with how a particular service is implemented. However, the user is interested in comparing the same service offered by different providers in terms of universal, user-oriented performance parameters. This implies that performance should be expressed by parameters that:

- Take into account all aspects of the service from the user's point of view.
- Focus on user-perceivable effects, rather than their causes within the network.
- Are independent of the specific network architecture or technology.
- Can be objectively or subjectively measured at the service access point.
- Can be easily related to network performance parameters.
- Can be assured to a user by the service providers(s).

On the other hand, there are NP (Network Performance) parameters that contribute towards QoS as experienced by the user. Network performance is defined as the ability of a network or network portion to provide the functions related to communications between users. Network performance may or may not be on an end-to-end basis. For example, access performance is usually separated from the core network performance in the operations of single IP-managed network, while Internet performance often reflects the combined NP of several autonomous networks.

The key NP parameters that impact the user are:

Delay: Delay manifests itself in a number of ways, including the time taken to establish a particular service from the initial user request and the time to receive specific information once the service is established. Delay has a very direct impact on user satisfaction depending on the application, and includes delays in the terminal, network, and any servers.

Note that from a user point of view, delay also takes into account the effect of other network parameters such as throughput.

Delay variation: Delay variation is generally included as a performance parameter since it is very important at the transport layer in packetized data systems due to the inherent variability in arrival times of individual packets. However, services that are highly intolerant of delay variation will usually take steps to remove (or at least significantly reduce) the delay variation by means of buffering, effectively eliminating delay variation as perceived at the user level (although at the expense of adding additional fixed delay).

Information loss: Information loss has a very direct effect on the quality of the information finally presented to the user, whether it is voice, image, video or data. In this context, information loss is not limited to the effects of bit errors or packet loss during transmission, but also includes the effects of any degradation introduced by media coding for more efficient transmission (e.g. the use of low bit-rate speech codecs for voice).

The performance considerations for different applications are provided in [75]. The recommendation provides also an indication of suitable performance targets for audio, video and data applications. Based on these requirements, the various applications can be mapped onto axes of packet loss and one-way delay as indicated in figure 8. The figure indicates the limit of delay and information loss tolerable for each indication class.

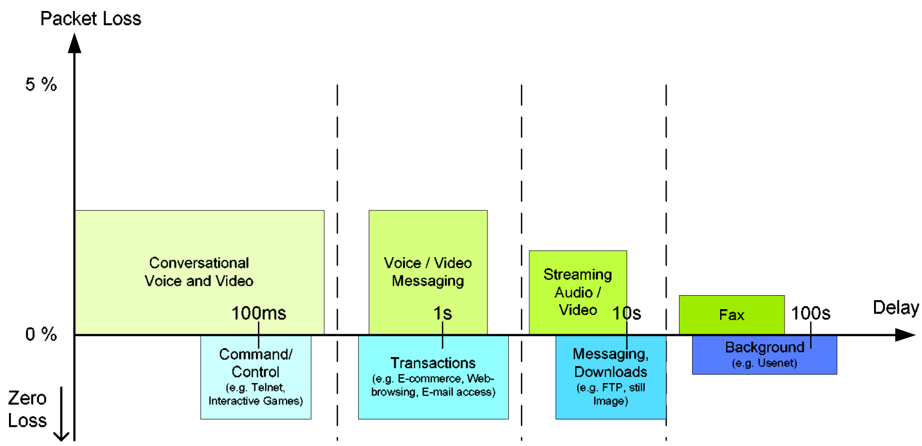

Figure 8: Mapping of user-centric QoS parameters

There are eight distinct groupings which encompass the range of applications identified. Within these eight groupings there is a primary segregation between applications that can tolerate some information loss and those that can not tolerate any information loss at all, and four general areas of delay tolerance.

This mapping is formalized in figure 9, to provide a recommended model for end-user QoS categories, where the four areas of delay are given names chosen to illustrate the type of user interaction involved. Each category could be subdivided into further categories to provide a range of quality levels for a specific service, as has been done for conversational voice in [76].

Error tolerant	**Conversational Voice and Video**	**Voice / Video Messaging**	**Streaming Audio / Video**	**Fax**
Error intolerant	**Command/Control** (e.g. Telnet, Interactive Games)	**Transactions** (e.g. E-commerce, Web-browsing, E-mail access)	**Messaging, Downloads** (e.g. FTP, still Image)	**Background** (e.g. Usenet)
	Interactive (delay << 1s)	Responsive (delay ~ 2s)	Timely (delay ~ 10s)	Non-critical (delay >> 10s)

Figure 9: Model for user-centric QoS categories

5.6.3.4 QoS activities in standardization bodies

5.6.3.4.1 ETSI EP TIPHON

ETSI EP TIPHON is working on end-to-end QoS issues (ETSI TR-STQ 037 - see bibliography) in a view of two aspects:

- call quality (related to a quality of media transmission);
- call set-up quality (characterized by the call-set up time as a principal parameter of signalling quality).

As far as concerns the call quality, ETSI EP TIPHON addresses three classes of end-to-end speech QoS [76]: WIDEBAND, NARROWBAND and BEST EFFORT. The TIPHON speech QoS classes WIDEBAND and NARROWBAND will provide performance guarantees for 95 % of all connections. The BEST EFFORT class provides no speech performance guarantees. The classes are defined from mouth-to-ear and therefore include both the network and the TIPHON terminal characteristics. Each of the classes is specified by three performance metrics: Overall Transmission Quality Rating (R), Listener Speech Quality (One-way non-interactive end-to-end Speech Quality) and End-to-end (mean one-way) Delay.

The current work on QoS in TIPHON is aimed at provisioning of a range of different levels of "guaranteed" end-end voice quality with the user being able to decide what quality to select either on a semi-permanent or per call basis. The five TIPHON speech classes are defined in [77]. Four of these classes involve achieving quality that is guaranteed to be above a certain level. Since calls are likely to traverse more than one network, the concept of guarantee makes it necessary for networks to exchange signalling about the quality that they can achieve, thus a substantial proportion of the current work in TIPHON is focussing on specifying the call QoS/QoS signalling.

The work on QoS for multimedia services has begun by definition of multimedia QoS classes [78]. The document is only partly complete. The TIPHON framework for multimedia QoS classes is given in figure 10.

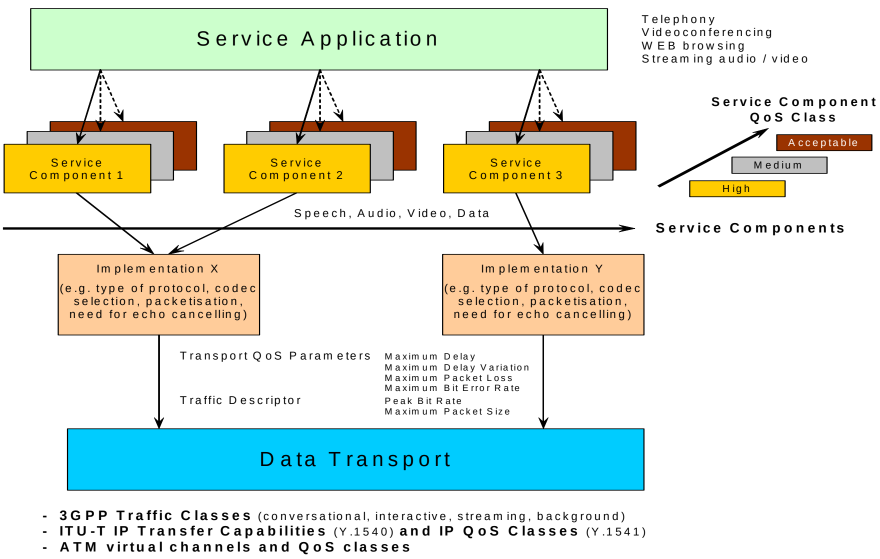

Figure 10: TIPHON framework for multimedia QoS standardization

The multimedia services are in general a combination of several service components, each with different QoS requirements. The following service components have been identified:

- **Speech**: narrowband voice telecommunication, focussing on interactive mouth to ear communication.
- **Audio**: wideband telecommunication of sound in general, focussing on acoustic fidelity.
- **Video**: telecommunication of full motion pictures, and of stills, focussing on visual fidelity.
- **Data**: telecommunication of data-files, focussing on error-free, and possibly timely, transfer.

The various service components listed above can be categorized according to the type of traffic they generate and the corresponding general performance characteristics. The following categories have been identified, based mainly on their delay sensitivity.

Table 8: General QoS characteristics for traffic categories

Traffic category	Service components	General QoS characteristics
Real-time conversational (e.g. telephony, teleconference, videophony, videoconference)	Speech Audio Video Multimedia	Delay sensitive Delay variation sensitive Limited tolerance to loss/errors (depending on coding) CBR (Constant Bit Rate) and VBR (Variable Bit Rate)
Real-time streaming (e.g. audio and video broadcast, surveillance, graphics)	Audio Video Multimedia	Tolerant to delay Delay variation sensitive (depending on buffer sizes in terminals/gateways) Limited tolerance to loss/errors (depends on coding) VBR
Near-real time interactive (e.g. web browsing)	Data	Delay sensitive (interactive services) Tolerant to delay variation Error sensitive VBR
Non-real time background (e.g. e-mail, file transfer)	Data	Not delay sensitive Not delay variation sensitive Error sensitive ABR

Table 9 indicates which service components might be applicable to a specific traffic category.

Table 9: Service components applicable to traffic categories

Service components	Traffic categories			
	Conversational	Streaming	Interactive	Background
Speech	X		X	
Audio	X	X		
Video	X	X		
Data			X	X

The TIPHON QoS category characteristics reflect the UMTS QoS service classes identified in clause 5.6.3.3.

5.6.3.4.2 ITU-T

There are several Study Groups within ITU-T that are involved in QoS issues.

ITU-T Recommendation SG 12 (End-to-end transmission performance of networks and terminals)

ITU-T Recommendation SG 12 is responsible for guidance on the end-to-end transmission performance of networks, terminals and their interactions, in relation to the perceived quality and acceptance by users of text, speech, and image applications. This work includes the related transmission implications of all networks (e.g. those based on PDH, SDH, ATM and IP) and all telecommunications terminals (e.g. handset, hands-free, headset, mobile, audiovisual, and interactive voice response). As lead ITU-T SG on QoS and performance, Study Group 12 provides leadership for the ITU-T in dealing with QOS-related issues. Internal to the ITU-T, this leadership involves providing a roadmap for QOS activities that can be used to identify and resolve QoS-related issues across Study Groups. External to the ITU-T, this leadership involves active communication with other organizations, with a goal of improving the visibility of ITU-T expertise in QoS and more effectively leveraging this expertise in specifications being developed elsewhere in the industry.

The recommendations directly related to QoS of multimedia services follow:

- **G.1000 [**74**] (Communication quality of service: A framework and definitions):** The recommendation addresses the need for a consistent approach to QoS, with a goal of setting a well-defined and relevant approach that can be readily used to plan and deploy networks, and to monitor service quality.

- **G.1010 [**75**] (End-user multimedia QoS categories):** The recommendation provides guidance on key factors that influence QoS from the perspective of the end-user. By considering a range of applications involving the media of voice, video, image and text, and the parameters that govern end-user satisfaction for these applications, a broad classification of end-user QoS categories is determined.

The current work areas directly related to QoS of multimedia services include:

- Question 10/12 (WP 2/12) - Transmission planning for voiceband, data and multimedia services.

- Question 13/12 (WP 3/12) - Multimedia QoS/performance requirements.

ITU-T Recommendation SG 13 (Multi-protocol and IP-based networks and their internetworking)

ITU-T Recommendation SG 13 is responsible for internetworking of heterogeneous networks encompassing multiple domains, multiple protocols and innovative technologies with a goal to deliver high-quality, reliable networking. Specific aspects are architecture, interworking and adaptation, end-to-end considerations, routing and requirements for transport. It is a lead SG for IP related matters, B-ISDN, Global Information Infrastructure and satellite matters.

Though there are no specifications directly related to QoS of multimedia services, the following ones are relevant in a view of specification of the network performance parameters:

Y.1540 (Internet protocol data communication service - IP packet transfer and availability performance parameters): This Recommendation [79] defines parameters that may be used in specifying and assessing the performance of speed, accuracy, dependability, and availability of IP packet transfer of international Internet Protocol (IP) data communication service. The defined parameters apply to end-to-end, point-to-point IP service and to network portions.

Y.1541 (Network performance objectives for IP-based services): This Recommendation [80] defines classes of network Quality of Service (QoS), and specifies provisional objectives for Internet Protocol network performance parameters. It specifies six different network QoS classes (Class 0, Class 1, Class 2, Class 3, Class 4, Class 5 - Unspecified). Y.1541 [80] applies to international end-to-end IP network paths. The QoS classes here are directly based on network performance parameters (IP packet transfer delay, delay variation, packet loss ratio and packet error ratio). The network QoS classes defined here are intended to be the basis of agreements between end-users and network service providers, and between service providers. Unlike the ETSI EP TIPHON, the ITU-T Recommendation Y.1541 [80] takes the approach to first define a set of network classes; in a second step the user QoS is guaranteed by choosing the appropriate network class for the application to be offered.

Y.1530 (Call processing performance for voice service in hybrid IP networks): This Recommendation defines performance parameters and objectives for point-to-point call processing in voice service for Hybrid IP networks.

ITU-SG 2 (Operational aspects of service provision, networks and performance)

ITU-SG 2 is responsible for studies relating to principles of service provision, definition and operational requirements of service emulation; numbering, naming, addressing requirements and resource assignment including criteria and procedures for reservation and assignment; routing and interworking requirements; human factors; operational aspects of networks and associated performance requirements including network traffic management, quality of service (traffic engineering, operational performance and service measurements); operational aspects of interworking between traditional telecommunication networks and evolving networks; evaluation of feedback from operators, manufacturing companies and users on different aspects of network operation.

Recommendations relevant to QoS aspects include mainly E. 8xx series. The current work areas related to QoS include:

- Question 2/2 (WP 1/2) - Routing and interworking plans for fixed and mobile networks.

- Question 5/2 (WP 5/2) - Service quality of networks.

Other ITU-T Study Groups involved in QoS issues include:

- **ITU-T Recommendation SG 4:** Management of QoS and SLA.
- **ITU-T Recommendation SG 9:** QoS for cable networks and video assessment.
- **ITU-T Recommendation SG 11:** QoS signalling.
- **ITU-T Recommendation SG 15:** System specific requirements for network and transport equipment.
- **ITU-T Recommendation SG 16:** QoS mechanisms for H.323-based multimedia systems. Quality of speech and video codecs.
- **ITU-T Recommendation SG 17:** Frame Relay QoS.

5.6.3.4.3 3GPP IP Multimedia

The QoS issues are studied also in 3GPP. Work on the specifications for 3GPP is divided into Releases. The first release that addresses the use of a packet (IP) network infrastructure for voice and multi-media services is Release 5 which contains the IP Multimedia development, whose specifications are currently being completed. Based on the ETSI TS 123 107 [81], the services are considered end-to-end, from a terminal equipment to another terminal equipment. An end-to-end service may have a certain QoS which is provided for the user of a network service. It is the user that decides whether he is satisfied with the provided QoS or not. To realize a certain network QoS a Bearer Service with clearly defined characteristics and functionality has to be set up from the source to the destination of a service. A bearer service includes all aspects to enable the provision of a contracted QoS. These aspects are among others the control signalling, user plane transport and QoS management functionality. Thus, it is the Bearer Service that provides the UMTS QoS.

UMTS defined four different QoS classes. It is necessary to be taken into account that when defining the UMTS QoS classes (also referred to as traffic classes), the restrictions of the air interface were taken into account. The main distinguishing factor between UMTS QoS classes is how delay sensitive traffic is. The UMTS QoS classes are given in table 10.

Table 10: UMTS QoS classes

Traffic class	Conversational class Real-time conversation	Streaming class Real-time stream	Interactive class Interactive best-effort	Background class Background best effort
Fundamental characteristics for QoS	Preserve time relation (variation) between information entities of the stream Conversational pattern (stringent and low delay)	Preserve time relation (variation) between information entities of the stream	Request response pattern Preserve payload content	Destination is not expecting the data within a certain time Preserve payload content
Example of the application	Voice	Streaming video	Web browsing	Background download of e-mails

The service provided by the UMTS network to the user of the UMTS bearer service is described by the means of the UMTS bearer service attributes. The service is then specified by a set of QoS attributes (QoS profile). At UMTS bearer service establishment or modification different QoS profiles have to be taken into account. A set of QoS attributes for each of four UMTS QoS is defined. It includes:

- Maximum bit/rate.
- Delivery order.
- Maximum SDU size in bytes.
- SDU format information.
- Delivery of erroneous SDUs.
- Residual BER.

- SDU error ratio.
- Transfer delay (ms).
- Guaranteed bit rate (kbit/s).
- Traffic handling priority.
- Allocation/retention priority.
- Source statistic descriptor.

A set of QoS attributes are defined also for the Radio Access Bearer Service.

5.6.3.4.4 ETSI IP Cablecom

ETSI's AT (Access and Terminals) Technical Committee is undertaking a programme of work adapting and developing further a set of specifications for IP-based cable networks. Initially the aim of IPCablecom is to support public telephony services with a quality that is at least as good as on the circuit switched networks. Later IPCablecom plans to support a range of multi-media services. IPCablecom is specified in a multipart standard ETSI TS 101 909 [82]. QoS is specified two parts:

- Dynamic Quality of Service for the Provision of Real Time Services over Cable Television Networks using Cable Modems (Part 5).
- Inter-domain Quality of Service (Part 17).

Different approaches are taken for the access and backbone networks. The access network uses IntServ techniques such as reservation where each session has its QoS managed individually. The backbone network uses DiffServ techniques were all communications that require similar treatment are handled a single class.

5.6.4 Security

5.6.4.1 Security aspects

In the telecommunication sector, security has become an increasingly important requirement for the various players, i.e. users, who require that their communications be kept confidential; network operators and service providers, who need to protect their activities and financial interests; and finally, regulatory bodies, which require and impose security measures by publishing directives and issuing regulations to ensure the availability of services (ITU-D IP Telephony Group of Experts Report -see bibliography). However, setting up a well-defined set of requirements in respect of security services remains a very difficult and fairly abstract notion, since each network has its own characteristics and because of the fact that security solutions depend on a variety of factors.

Call protection can be regarded from two points of view:

User point of view: It is on the users themselves to ensure the protection of their calls. In this case, the public network does not intervene. This type of protection is known as end-to-end protection

Network point of view: It consists in wholly or partly delegating responsibility for the protection of calls to the public network, which must ensure protection on individual portions of the network located between two sets of public network security equipment.

The most important features characterizing network security are the following:

Confidentiality, whereby a call between two correspondents is protected against illegal tapping by an unauthorized or ill-intentioned third party.

Authentication, whereby an entity can be sure that the data received actually come from the stated transmitting entity.

Access control, which is a service whereby access to the network resources (server, switch, router, etc.) is restricted in accordance with the security policy in force. Otherwise, if an ill-intentioned individual manages to obtain unauthorized access to one of the network resources, he or she is then able to launch attacks such as illegal tapping or denial of service, which consists in the continuous transmission of data to network elements such that no resources remain available for other network users.

Integrity, whereby an entity can be sure that the data received have not been modified in any way during the course of their transfer. It is possible, by means of this service, to eliminate the risk of data corruption as a result of deliberate and ill-intentioned manipulation.

5.6.4.2 Security in PSTN/ISDN

One of the main differences between the PSTN/ISDN and IP networks lies in the concentration of intelligence and in processing within the network at the level of switching nodes. In the case of the PSTN/ISDN, protection is fully the responsibility of the network. The fact that the intelligence is located within the switches considerably reduces the risk of ill-intentioned attacks. However, the PSTN/ISDN is today totally free from criminal activity or hacking. The development and introduction of the intelligent network using SS7 (Signalling System No.7) provided on one hand greater flexibility to the network through the introduction of new services, on the other hand it increased its vulnerability to the misuse of those services, an example of which is the freephone service. Certain services, moreover, are more open to misuse since their utilization requires that users have access to management information.

As regards the security features referred to above, the following ones may be pointed out in respect of the telephone network:

Confidentiality: The PSTN/ISDN provides full confidentiality that is limited solely by the legislation in force (tapping of a telephone line by the national authorities).

Authentication: A call can only be established if the calling party can be properly identified by the network, such authentication being an essential factor in service billing. With the PSTN/ISDN network, it is therefore possible at all times to know the two parties to a call (i.e. the called and calling parties).

Access control: The switches are generally housed in well-protected locations (telephone exchanges), an access control system can be designed to minimize the risk of attack by an anonymous individual. Furthermore, switches generally allow for the safeguarding of all actions initiated from a maintenance console, access to which is in the majority of cases protected by passwords.

Integrity: The circuit switching used in PSTN/ISDN, whereby a 64 kbit/s circuit is reserved throughout the duration of a call, facilitates the task of ensuring call integrity.

5.6.4.3 Security in IP networks

In IP networks, most of the processing needed to establish calls is delegated to the user terminal equipment. The intelligence is hence deployed to the ends rather than to the nodes of the network, as in the case of traditional telecommunication networks. Thus, the security functions as well will to a large extent be ensured by users, and, as the case may be, by the end routers, and not by core equipment within the network. In a managed IP network, the management, maintenance and operating functions are the responsibility of a well-identified party. In this case, the network manager could introduce protocols and equipment for the purpose of implementing security services within the network, whereupon the responsibility of ensuring communication security would be borne in part by the network. In the Internet, which is, in fact, the interconnection of a very large number of IP networks worldwide, the lack of any party having overall responsibility for this "network of networks" requires that users take full responsibility for ensuring the security of their communications.

Furthermore, security issues were not taken into account at the design stage of the IP protocol. This is why it has been necessary, in the interests of ensuring the protection of communications carried over such networks, to subsequently add security services to the mass of network protocols already in existence. Two solutions have prevailed for the security of traffic transported over IP, namely the TLS (Transport Level Security) protocol, which provides security within the transport layer, and the IPSec protocol. The TLS protocol is implemented on top of the TCP and is therefore only able to protect applications traffic transported over TCP, whereas IPSec is applied at the IP level and is hence more generic than TLS and can be used to provide security for any type of traffic over IP.

There are two modes for the provision of security for IP packets using IPSec: the transport mode and the tunnel mode:

The transport mode applies one or more security functions (essentially authentication and encryption) to the IP packet to be transmitted. These functions do not provide complete protection for the header fields. The mode of transport is applicable only to terminal equipment, particularly the end routers. An intermediate router might not apply the IPSec transport mode to an IP packet it is relaying on account of fragmentation and reassembly problems.

In **tunnel mode**, a new IP packet is created by a method involving the opening of an IP tunnel in IP. The security function(s) which is (are) applied to the external IP packet therefore protect the integrity of the original internal IP packet (header and data), since this constitutes the "data" part of the external packet. This is obviously the best mode for creating secure VPNs and ensures better protection against traffic flow analysis.

5.6.4.4 Security for multimedia

In terms of multimedia services, security aspects become more complex, especially for two reasons:

- specific nature of multimedia communication (diverse media, multiple streams within one communication);
- the use of multimedia communications in the course of services and applications in new key areas such e-business, e-government, e-health and e-learning. While the mentioned applications put even greater emphasis on the objectives of confidentiality, integrity, and non-repudiation of communications, they also raise new issues of protection of intellectual property distributed over the telecommunications network.

Security for multimedia services becomes increasingly important and includes a wide range of issues, like end-to-end privacy of data, authentication (user identification), anonymous access, access control intrusion detection, electronic signature, encryption, non-repudiation, lawful intercept.

5.6.4.5 Activities on security issues in standardization bodies

5.6.4.5.1 ITU-T

ITU-T Recommendation SG 16 (Multimedia services, systems and terminals)

ITU-T Recommendation SG 16 is aimed at security aspects and mechanisms for H.323-based multimedia systems. Recommendations relevant to security aspects include the following recommendations:

- H.233 (Confidentiality system for audiovisual services) [83].
- H.234 (Encryption key management and authentication system for audiovisual services) [84].
- H. 235 (Security and encryption for H-series multimedia terminals) [85].
- H.323 annex J (Packet based communication systems - Security for simple endpoint types).

The security aspects for facsimile are covered in ITU-T Recommendations T.30 (annex G and annex H) [86], T.36 [87], T.503 [88] and T.563 [89].

The current work area related to security of multimedia systems and services is under the responsibility of the WP2/16 (Multimedia platform and interworking).

ITU-T Recommendation SG 17 (Data networks and telecommunication software)

ITU-T Recommendation SG 17 is responsible for studies relating to data communication networks, application of open system communication including networking, directory and security, technical languages, the method for their usage and other issues related to the software aspects of telecommunication systems. It is a lead Study Group on FR (Frame Relay), communications system security and languages and description techniques.

Recommendations on security issues under the responsibility of the SG 17 may be grouped to several areas:

- Security architecture framework (recommendations X.800 [90], X.802 [91] and X.803 [92] describe security within the context of open systems, X.810 [93], X.811 [94], X.812 [95], X.813 [96], X.814 [97], X.815 [98] and X.816 [99] provide security frameworks covering aspects of security such as authentication, access control, non-repudiation, confidentiality, integrity, and security audit and alarms).

- Security techniques (X. 841, X.842 [101] and X.843 [102] recommendations on security information objects and trusted third party services have been established in co-operation with ISO/IEC JTC 1/SC 27).
- Security protocols (X.273 [103], X.274 [104]).
- Directory services & authentication (X.500 [105], X.501 [106], X.509 [107], X.519 [108]).
- Systems management (X.733 [109], X.735 [110], X.736 [111], X.740 [112], X.741 [113]).
- Security in Frame Relay (X.272 [114]).

The current work area related to security requirements, models and guidelines for communication systems and services is under the responsibility of the Working Party 2/17 (Open Systems Technology), and is focused on new security requirements and their solutions. In the initial approach, four principal areas on security requirements on communication systems security, security management, mobile security and tele-biometric systems have been identified.

Other ITU-T Study Groups involved in security issues include:

- ITU-T Recommendation SG 4 (Telecommunication management, including TMN): Network management security.
- ITU-T Recommendation SG 9 (Integrated broadband cable networks and television and sound transmission): Televisions and cable systems.

ITU-T special project MEDIACOM 2004

The prime focus of the security part of the project is about all aspects of multimedia security. The overall aspect of multimedia security has been grouped into several categories:

- Security requirements and security services.
- Security architecture and infrastructure.
- Multimedia communication security.
- Interdomain security.
- Management and usability.
- Security standards.

5.6.4.5.2 ETSI

There is actually no TB (Technical Body) within ETSI directly related to security aspects of multimedia services. However, several TBs are working on security issues.

ETSI SG SAGE (Special Committee Security Algorithms Group of Experts)

The Security Experts Group is responsible for creating ETSI reports (containing confidential specifications), draft I-ETSs and ETSs in the area of cryptographic algorithms and protocols specific to fraud prevention/unauthorized access to public/private telecommunications networks and user data privacy.

ETSI EP TIPHON

Work on security is going within the ETSI EP TIPHON WG 8. Recently, two ETSI TSs have been published:

- ETSI TS 102 165-1 [115]. The TS derives, by means of a threat analysis, the requirements for security features that when implemented are necessary and sufficient to ensure that TIPHON derived products do not harm to their participants.
- ETSI TS 102 165-2 [116]. The TS defines by means of meta-protocol, algorithm boundary conditions, and guidance text the security countermeasures identified in ETSI TS 102 165-1 [115].

The current work is focused on analysis of IPv6 application in telecommunications standards and evaluation criteria for cryptographic algorithms.

ETSI TC SEC (Security Group)

It has been recently created within ETSI. The group's primary role is to provide a light-weight horizontal co-ordination structure for security issues. TC SEC has released a draft technical report *XML Format for Signature Policies* as a Final Phase 3 deliverable. ESI (Electronic Signature and Infrastructure) Working Group of ETSI SEC and Specialist Task Forces are acting in co-operation with CEN/ISSS within the ITCSB/EESSI work programme.

TC SEC is also in the process of producing a technical specification ETSI TS 101 903: *XML Advanced Electronic Signatures (XAdES)*) that defines a XML format for electronic signatures compliant with the European Directive.

5.6.4.5.3 IETF

There is the Security Area within the IETF that is responsible for development of security oriented protocols, security review of RFCs, development of candidate policies, and review of operational security on the Internet. Within the Security Area, several Working Groups have been active in defining security protocols and infrastructure facilities. Some current and recently active IETF Security Area working groups include:

PKIX

Public-Key Infrastructure (X.509), profiling usage of X.509 certificates and CRLs and defining associated PKI protocols (e.g. certificate management, certificate validation). A PKI (Public-Key Infrastructure) consists of protocols, services, and standards supporting applications of public-key cryptography. PKI sometimes refers simply to a trust hierarchy based on public-key certificates and in other contexts embraces encryption and digital signature services provided to end-user applications as well. A middle view is that a PKI includes services and protocols for managing public keys, often through the use of CA (Certification Authority) and RA (Registration Authority) components, but not necessarily for performing cryptographic operations with the keys.

Among the services likely to be found in a PKI are the following:

- Key registration - issuing a new certificate for a public key.
- Certificate revocation - cancelling a previously issued certificate.
- Key selection - obtaining a party's public key.
- Trust evaluation - determining whether a certificate is valid and what operations it authorizes.

Key recovery has also been suggested as a possible aspect of a PKI. There is no single pervasive public-key infrastructure today, though efforts to define a PKI generally presume there will eventually be one, or, increasingly, that multiple independent PKIs will evolve with varying degrees of coexistence and interoperability. In this sense, the PKI today can be viewed akin to local and wide-area networks in the 1980's, before there was widespread connectivity via the Internet. As a result of this view toward a global PKI, certificate formats and trust mechanisms are defined in an open and scaleable manner, but with usage profiles corresponding to trust and policy requirements of particular customer and application environments. Efforts to define a PKI today are underway in several governments as well as standards organizations.

IPSec

IPSec Working Group is defining a set of specifications for cryptographically-based authentication, integrity, and confidentiality services at the IP datagram layer. IPSec is intended to be the future standard for secure communications on the Internet, but is already the de facto standard. The IPSec group's results comprise a basis for interoperable secured host-to-host pipes, encapsulated tunnels, and Virtual Private Networks (VPNs), thus providing protection for client protocols residing above the IP layer.

The protocol formats for IPSec's AH (Authentication Header) and IP ESP (Encapsulating Security Payload) are independent of the cryptographic algorithm, although certain algorithm sets are specified as mandatory for support in the interest of interoperability. Similarly, multiple algorithms are supported for key management purposes (establishing session keys for traffic protection), within IPSec's IKE (Internet Key Exchange) framework.

The relevant RFC documents and Internet-Drafts can be found at: http://www.ietf.org/html.charters/ip sec-charter.html.

S/MIME (Secure/Multipurpose Internet Mail Extensions)

S/MIME is a protocol that adds digital signatures and encryption to Internet MIME (Multipurpose Internet Mail Extensions) messages described in RFC 1521. MIME is the official proposed standard format for extended Internet electronic mail. Internet e-mail messages consist of two parts, the header and the body. The header forms a collection of field/value pairs structured to provide information essential for the transmission of the message. The structure of these headers can be found in RFC 822 [117]. The body is normally unstructured unless the e-mail is in MIME format. MIME defines how the body of an e-mail message is structured. The MIME format permits e-mail to include enhanced text, graphics, audio, and more in a standardized manner via MIME-compliant mail systems. However, MIME itself does not provide any security services. The purpose of S/MIME is to define such services, following the syntax given in PKCS #7 (Public Key Cryptography Standards, PKCS#7 defines a general syntax for messages that include cryptographic enhancements such as digital signatures and encryption) for digital signatures and encryption. The MIME body section carries a PKCS #7 message, which itself is the result of cryptographic processing on other MIME body sections. S/MIME standardization has transitioned into IETF, and a set of documents describing S/MIME version 3 have been published there.

PGP/MIME (Pretty Good Privacy/Multipurpose Internet Mail Extensions) provides also for digital signature and encryption of MIME based messages, but is based on the PGP (Pretty Good Privacy) format programs. Although primary application is e-mail, it can be used also in other context.

TLS (Transport Layer Security)

Transport Layer Security, defining the standardized successor to the widely-deployed SSL (Secure Sockets Layer) protocol. The SSL Handshake Protocol was developed by Netscape Communications Corporation to provide security and privacy over the Internet. The protocol supports server and client authentication. The SSL protocol is application independent, allowing protocols like HTTP (HyperText Transfer Protocol), FTP (File Transfer Protocol), and Telnet to be layered on top of it transparently. Still, SSL is optimized for HTTP; for FTP, IPSec might be preferable. The SSL protocol is able to negotiate encryption keys as well as authenticate the server before data is exchanged by the higher-level application. The SSL protocol maintains the security and integrity of the transmission channel by using encryption, authentication and message authentication codes.

TLS (Transport Layer Security) is a protocol that is based on and very similar to SSL 3.0 [118].

The WTLS (Wireless TLS) specifies the security layer protocol in WAP (Wireless Application Protocol). It is very similar to TLS but optimized for low-bandwidth bearer networks.

CAT

Common Authentication Technology, defining mechanisms and interfaces GSS-API (Generic Security Service API) for callable integration of security services into applications. The GSS-API is a CAPI (Cryptography Application Programming Interface) for distributed security services. It has the capacity to handle session communication securely, including authentication, data integrity, and data confidentiality. The GSS-API is designed to insulate its users from the specifics of underlying mechanisms. GSS-API implementations have been constructed atop a range of secret-key and public-key technologies. The current Version 2 GSS-API definition is available in the RFC 2078.

XMLDSIG (XML Digital Signatures)

XML Digital Signatures have been chartered in conjunction with the World-Wide Web Consortium to define digital signature facilities for XML documents.

SPKI (Simple Public-Key Infrastructure)

SPKI has issued experimental documents concerning definition and usage of certificates in a non-X.509 format.

OPENPGP (Open Specification for Pretty Good Privacy)

An OPENPGP defines a specification for message and key formats as used in PGP (Pretty Good Privacy). PGP is a software package that provides cryptographic routines for e-mail and file storage applications. It is based on existing cryptosystems and cryptographic protocols, but was developed to run on multiple platforms. It provides message encryption, digital signatures, data compression, and e-mail compatibility.

SSH (Secure Shell)

SSH defines specifications for the Secure Shell protocol. It is a protocol which permits secure remote access over a network from one computer to another. SSH negotiates and establishes an encrypted connection between an SSH client and an SSH server, authenticating the client and server in any of a variety of ways (some of the possibilities for authentication are RSA (Rivest, Shamir and Adleman), SecurID, and passwords). That connection can then be used for a variety of purposes, such as creating a secure remote login on the server (effectively replacing commands such as telnet, rlogin, and rsh) or setting up a VPN (Virtual Private Network).

KRB (Kerberos)

Kerberos is a network authentication protocol. It provides a means of verifying the identities of principals on an open network without relying on assertions by the host operating system, without basing trust on host addresses, without requiring physical security of all the hosts on the network, and under the assumption that packets travelling along the network can be read, modified, and inserted at will. Kerberos performs authentication under these conditions as a trusted third-party authentication service by using conventional (shared secret key) cryptography. A key characteristic of Kerberos is that the protocol is designed to operate across organizational boundaries. By establishing "inter-realm" keys, the administrators of two realms can allow a client authenticated in the local realm to use its authentication remotely. RFC 1510: "The Kerberos Network Authentication Service (V5) [119] provides an overview and specification of Kerberos Version 5.

Kerberos provides a general mechanism for protocol extensibility. Kerberos extensions (outside the scope of RFC 1510 [119]) can, for example, provide for the use of public key cryptography during certain phases of the authentication protocol. Such extensions support Kerberos authentication for users registered with public key certification authorities and provide certain benefits of public key cryptography in situations where they are needed.

DNS Security

DNS security issues are under the responsibility of the dnsext (DNS Extensions) WG. DNS Security documents fall into one or possibly more of the following sub-categories: new DNS security resource records, implementation details of specific digital signing algorithms for use in DNS Security and DNS transaction authentication. Recently, a new Internet Draft "DNS Security Document Roadmap" (DNS Security Document Roadmap - see bibliography) has been published. DNS Security (DNSSEC) technology is composed of extensions to the DNS protocol that provide data integrity and authentication to security aware resolvers and applications through the use of cryptographic digital signatures). The main goal of the DNSSEC extensions is to add data authentication and integrity services to the DNS protocol. These protocol extensions should be differentiated from DNS operational security issues. Since the goal of DNS Security extensions is to become part of the DNS protocol standard, additional documents that seek to refine a portion of the security extensions will be introduced as the specifications progress along the IETF standards track.

Intrusion Detection

Intrusion detection issues are under the responsibility of the Intrusion Detection Working Group. Its purpose is to define data formats and exchange procedures for sharing information of interest to intrusion detection and response systems, and to management systems which may need to interact with them.

5.6.4.5.4 ISO

There is the JTC1/SC27 "IT Security Techniques" Special Committee that is responsible for IT security aspects within ISO. Three Working Groups are currently active under the JTC1/SC27:

- JTC1/SC27/WG1: "Requirements, security services and guidelines".
- JTC1/SC27/WG2: "Security techniques and mechanisms".
- JTC1/SC27/WG3: "Security evaluation criteria".

The standards cover the following areas:

Framework and common criteria for IT security standards:

ISO/IEC 17799:2000: "Information technology - Code of practice for information security management" [120] is intended for use as a reference document by those who are responsible for developing, implementing and maintaining information security within their organization.

ISO/IEC 15408:1999: "Information technology - Security techniques - Evaluation criteria for IT Security" [121].

ISO/IEC WD 15443: "Information technology - Security techniques - A framework for IT security assurance".

ISO/IEC WD 15446: "Information technology - Security techniques - Guide on the Production of Protection Profiles and Security Techniques".

ISO/IEC: 15946:2002:"Information technology - Security techniques - Cryptographic techniques elliptic curves" [122]. The standard is based on FIPS PUB 140-2 "Security Requirements for Cryptographic Modules" which specifies the security requirements that are to be satisfied by a cryptographic module utilized within a security system protecting unclassified information within computer and telecommunication systems (including voice systems). The security requirements in FIPS 140-2 cover areas related to the secure design and implementation of a cryptographic module. These areas include basic design and documentation, module interfaces, authorized roles and services, physical security, software security, operating system security, key management, cryptographic algorithms, electromagnetic interference/electromagnetic compatibility (EMI/EMC), and self-testing.

ISO/IEC TR 13335: "Information technology - Guidelines for the management of IT Security (GMITS)" [123]. It is technical report providing guidance on the effective management of IT security.

ISO/IEC TR 15947: "Information technology - Security techniques - IT intrusion detection techniques" [124] defines a framework for detection of intrusions into IT systems. It seeks to establish common definitions for intrusion detection terms and concepts. It describes the methodologies and concepts and the relationships among them, addresses possible orderings of intrusion detection tasks and related activities, and attempts to relate these tasks and processes to an organization's or enterprise's procedures to demonstrate the practical integration of intrusion detection within an organization or enterprise security policy.

ISO/IEC 10181: "Information technology - Open Systems Interconnection - Security framework for open systems" [125] defines means of providing protection for open systems, objects within such systems, and interactions between systems.

ISO/IEC 11586: "Information technology - Open Systems Interconnection - Generic upper layers security" [126] defines a GULS (Generic Upper-Layers Security) security exchange protocol. The model includes a service definition, a protocol specification and a PICS pro-forma for ASEs (Application Service Elements) to support the provision of security services at the application layer level of OSI (Open System Interconnection).

ISO/IEC 15816 [127]/ITU-T Recommendation X.841 [100]: "Information technology - Security techniques - Security information objects for access control" provides object definitions that are needed in more than one security standard to avoid multiple and different definitions of the same functionality. It references existing definitions in other International Standards. The document contains methods and guidelines for defining basic security-related information objects and for constructing new ones from existing components. It also provides a collection of generic and specific SIO (Security Information Object) definitions.

Digital Signature and Encryption:

ISO/IEC 11770: "Information technology - Security techniques - Key management" [128]. The PKCS suite of specifications developed at RSA laboratories provides a wide range of algorithm-specific and algorithm-independent key cryptography specifications, including the Diffie-Hellman key exchange mechanisms that are widely used on both the Internet and other private and open networks. PKCS also defines an algorithm-independent syntax for digital signatures, digital envelopes (for encryption) and extended certificates that enable someone implementing any cryptographic algorithm whatsoever to conform to a standard syntax, and thus achieve interoperability.

To date the following PKCS specifications have been published:

- PKCS#1 - defines mechanisms for encrypting and signing data using the RSA public-key cryptosystem.
- PKCS#3 - defines a Diffie-Hellman key agreement protocol.
- PKCS#5 - describes a method for encrypting a string with a secret key derived from a password.
- PKCS#6 - describes a format for extended certificates; an extended certificate consists of an X.509 certificate together with a set of attributes signed by the issuer of the certificate (PKCS #6 is being phased out in favour of Version 3 of X.509).
- PKCS#7 - defines a general syntax for messages that include cryptographic enhancements such as digital signatures and encryption.

- PKCS#8 - describes a format for private-key information. This information includes a private key for some public-key algorithm, and optionally a set of attributes.
- PKCS#9 - defines selected attribute types for use in the other PKCS standards.
- PKCS#10 - describes a syntax for certification requests.
- PKCS#11 - defines a technology-independent programming interface, called Cryptoki, for cryptographic devices such as smart cards and PCMCIA cards.
- PKCS#12 - defines a *Personal Information Exchange Syntax Standard*.
- PKCS#13 - defines an *Elliptic Curve Cryptography Standard* (Draft).
- PKCS#15 - defines a *Cryptographic Token Information Format*.

ITU-T Recommendation X.509 [107]/ISO/IEC 9594-8:2000 [129]: "The Directory: Authentication framework". It describes two levels of authentication: simple authentication, based on use of a password to verify user identity, and strong authentication, using credentials formed using cryptographic techniques. The standard recommends that only strong authentication be used as the basis of providing secure services. The protocol used by applications to obtain credentials is the Directory Access Protocol (ITU-T Recommendation X.519 [108]/ISO/IEC 9594-5). The X.509 [107] standard also provides a means by which tokens can be signed within the Directory, but does not include a specific standard for digital signatures. The standard allows for one, two and three way authentication. Public-key cryptography is used for strong authentication, but the authentication framework is not dependent on the use of a particular cryptographic algorithm, though two users wishing to authenticate must support the same algorithm. The RSA cryptosystem is defined as an informative annex to the standard. The X.509 PKIX (Internet Public Key Infrastructure) specifications profile the format and semantics of X.509 certificates and certificate revocation lists for the Internet. Procedures are described for processing certification paths in the Internet environment. Operational protocols are provided for the most commonly used interchange formats, including those related to directories.

ISO 8372:1987: "Information processing - Modes of operation for a 64-bit block cipher algorithm" [130] specifies four modes of operation for a 64-bit block cipher algorithm.

ISO/IEC 10116: "Information technology - Security techniques - Modes of operation for an n-bit block cipher algorithm" [131] specifies four modes of operation for an n-bit block cipher algorithm.

ISO/IEC 9796: "Information technology - Security techniques - Digital signature scheme giving message recovery" [132] provides a digital signature scheme that includes:

- a signature process using a secret signature key and a signature function for signing messages;
- a verification process using a public verification key and a verification function for checking signatures while recovering messages.

ISO/IEC 9797: "Information technology - Security techniques - Message authentication codes (MACs)" [133] are data integrity mechanisms that compute a short string (the Message Authentication Code or MAC) as a complex function of every bit of the data and of a secret key. Their main security property is unforgiving: someone who does not know the secret key should not be able to predict the MAC on any new data string.

ISO/IEC 9979: "Information technology - Security techniques - Procedures for the registration of cryptographic algorithms (2nd edition)" [134] describes the role of the Registration Authority responsible for the maintenance of the ISO Register of cryptographic algorithms and the publishing of Register entries. ISO/IEC 9979 [134] also defines the three types of cryptographic algorithm to be registered:

- algorithms in which the complete description of the process accompanies the registration entry;
- algorithms in which the complete description of the process is defined in an ISO document, or a standard maintained by a Member Body of ISO or by a liaison organization;
- algorithms in which the complete description is not fully defined (or not defined at all).

The ISO Register of cryptographic algorithms (attachment 2 of ISO/IEC 9979 [134]) serves as a common reference point for the identification of cryptographic algorithms by a unique name. The register is also a repository of basic parameters identified with the register entry. The principal purpose of the register is to enable entities to identify and negotiate an agreed cryptographic algorithm.

ISO/IEC 10118: "Information technology - Security techniques - Hash-functions" [135] define hash-functions that map arbitrary strings of bits to a given range. They can be used for reducing a message to a short imprint for input to a digital signature mechanism, or for committing the user to a given string of bits without revealing this string. The input string of a hash- function is called a data string and the output string is called a hash-code.

ISO/IEC 11770: "Information technology - Security techniques - Key management" [128] provides procedures for handling cryptographic keying material to be used in a symmetric or asymmetric cryptographic algorithm according to the security policy in force.

ISO/IEC 13888: "Information technology - Security techniques - Non-repudiation" [136] defines a repudiation service, which role is to collect, maintain, make available, and validate evidence concerning a claimed event or action in order to resolve disputes about the occurrence or non-occurrence of the event or action. The event or act on can be the generation of a message, sending of a message, receipt of a message, or submission of a message transport of the message.

ISO/IEC 14888: "Information technology - Security techniques - Digital signatures with appendix" [137] makes use if collision-resistant hash functions, which are executed both in the signature and verification process. The other main function in the signature process are pre-sign and sign, where the pre-sign function is independent of the message and the sign function is determined by the signature key. In the verification process the main function is the verify function, which is determined by the verification key. It forms part of the recommendations from the EESSI (European Electronic Signature Standardization Initiative).

ISO/IEC 15946: "Information technology - Security techniques - Cryptographic techniques based on elliptic curves" [122] is a specialist form of cryptography defined over finite fields of prime power order (including the special cases of prime order and characteristic two). The representation of elements of the underlying finite field (i.e. which basis is used) is however outside the scope of this standard. The standard specifies public key cryptographic techniques based on elliptic curves. They include the establishment of keys for secret-key systems and digital signature mechanisms.

ISO/IEC TR 14516 [138]/ITU-T Recommendation X.842 [101]: "Information technology - Security techniques - Guidelines for the use and management of Trusted Third Party services" is a standard associated with the provision and operation of a TTP (Trusted Third Party). It issues a number of security related aspects, provides guidance for the use and management of TTPs, a definition of the basic duties and services provided, their description and their purpose, and the roles and liabilities of TTPs and entities using their services. The document identifies different major categories of TTP services including: time stamping, non-repudiation, key management, certificate management, and electronic notary public. Each of these major categories consists of several services which logically belong together. The document is intended primarily for system managers, developers, TTP operators and enterprise users to select those TTP services needed for particular requirements, their subsequent management, use and operational deployment, and the establishment of a Security Policy within a TTP.

ISO/IEC 15945 [139]/ITU-T Recommendation X.843 [102]: "Information technology - Security techniques - Specification of TTP services to support the application of digital signatures" defines those TTP services needed to support the application of digital signatures in commercial applications. It also defines interfaces and protocols to enable interoperability between entities associated with these TTP services.

5.6.5 Network management

Network management can be presented as a set of tools, applications and devices to assist network managers in monitoring and maintaining networks. The main goals of the network management are to provide higher network availability, reducing network operational cost, reducing network bottlenecks, increase flexibility of operation and integration, higher efficiency, easy of use and security.

5.6.5.1 ISO network management model

The ISO network management model [140] is the primary means for understanding the major functions of network management systems. The model consists of five conceptual areas:

- Fault management.
- Configuration management.
- Accounting management.
- Performance management.

- Security management.

This FCAPS (Fault, Configuration, Accounting, Performance, Security) model has become the major functionality criterion for evaluating the capability of various network management products. Fault management involves detection, isolation using analysis, and correction of unusual operational behaviour of systems in OSI environment. Configuration management is used to locate the resources, including the failed ones, and also to keep the track of the types of resources and their details. Accounting Management incorporates informing relevant users/authorities about the usage of resources and the associated costs. Performance Management is concerned with the behaviour and evaluation of the effectiveness of resources. Security Management involves network and network systems security.

Some other areas covered by the ISO network management model include:

- Chargeback - provides the means to charge the end-user for only the specific portion of the service he uses.
- Systems Management - management and administration of services provided on the network.
- Cost Management - addresses the reliability, operability and maintainability of managed objects.

5.6.5.2 Network Management protocols

SNMP (Simple Network Management Protocol)

Network Management protocols started with SNMP in 1988. The protocol was meant to handle IP based networks. All versions (SNMPv1, SNMPv2, and SNMPv3) of the Internet-Standard Management Framework share the same basic structure and components. Furthermore, all versions of the specifications of the Internet-Standard Management Framework follow the same architecture. An enterprise deploying the Internet-Standard Management Framework contains the following four basic components:

- Several managed nodes, each with an SNMP entity that provides remote access to management instrumentation (traditionally called an agent).
- At least one SNMP entity with management applications (traditionally called a manager).
- A management protocol used to convey management information between the SNMP entities.
- Management information.

This basic structure is common to all versions of the Internet-Standard Management Framework. The specifications of the Internet-Standard Management Framework are based on a modular architecture. The framework consists of:

- a data definition language;
- definitions of management information (MIB - Management Information Base);
- a protocol definition; and
- security and administration.

Over time, as the Framework has evolved from SNMPv1, through SNMPv2, to SNMPv3, the definitions of each of these architectural components have become richer and more clearly defined, but the fundamental architecture has remained consistent.

The original Internet-Standard Network Management Framework (SNMPv1) consists of three documents:

- RFC 1155 [141] - defines the SMI (Structure of Management Information) - the mechanisms used for describing and naming objects for the purpose of management.
- RFC 1212 [142] - defines a more concise description mechanism, but is wholly consistent with the SMI.
- RFC 1157 [143] - defines SNMP, the protocol used for network access to managed objects.

The SNMPv2 Management Framework is fully described in RFCs 1902 - 1907. Coexistence and transition issues relating to SNMPv1 and SNMPv2 are discussed in RFC 2576. It provides several advantages over SNMP v1, on the other hand, it is incomplete in that it does not include provision of security and administration.

The SNMPv3 Management Framework, as described in RFCs 2570 - 2575 (SNMP v3 charter - see bibliography) addresses the deficiencies in SNMPv2 relating to security and administration. Coexistence issues relating to SNMPv1, SNMPv2, and SNMPv3 can be found in RFC 2576.

The SNMPv3 RFCs were produced by the SNMPv3 IETF WG. It reused the SNMPv2 Draft Standard documents (i.e. RFCs 1902 through 1908). As a result, SNMPv3 is SNMPv2 plus security and administration. The new features of SNMPv3 in security administration issues include:

- Security (authentication and privacy, authorization and access control).
- Administrative Framework (naming of entities, people and policies, usernames and key management, notification destinations, proxy relationships, remotely configurable via SNMP operations).

CMIP (Common Management Information Protocol)

CMIP was defined by ISO for OSI systems. It was proposed as a standard to supersede SNMP. Although CMIP was more complex, SNMP remained in use and proliferated as Internet expanded.

TMN (Telecommunication Management Network)

TMN ITU-T Recommendation M.3000 Recommendation series was built over the OSI reference model but it includes support for SS7, TCP/IP, ISDN, X.25, 802.3 LAN based networks. It provides a framework for achieving interconnectivity and communication across heterogeneous operating systems and telecommunication networks. TMN principles are incorporated into a telecommunication network to send and receive information and to manage its resources (NEs - Network Elements in TMN terminology). TMN enables communication between OSS (Operation Support Systems) and NEs.

TMN uses object-oriented principles and standard interfaces to define communication between entities in a network. The standard management interface for TMN is called the Q3 interface. TMN architecture and interfaces build on existing ISO OSI standards, especially CMIP, GDMO (Guideline for Definition of Managed Objects), ASN.1 (Abstract Syntax Notation), OSI RM (Open System Interconnection Reference Model).

TMN is based on the OSI management framework and uses an object-oriented approach, with managed information in network resources modelled as attributes in managed objects. Management functions are performed by operations comprised of CMIS (Common Management Information Service) primitives. A network's managed information and the rules by which that information is presented and managed, is referred to as the MIB (Management Information Base). Processes that manage the information are called management entities. The TMN management structure is composed of four layers: Element Management, Network Management, Service Management and Business Management Layers.

Since their publication, the TMN standards have been embraced by other standards bodies, like ETSI, NMF (Network Management Forum - which developed e.g. TMN/C++ API to help ease the agent development process), Bellcore.

LMMP (LAN Man Management Protocol)

LMMP [144] was developed as a network management solution for LANs, i.e. over the IEEE 802 LLC (Logical Link Layer) [145]. It is limited to only the LAN boundary.

ANMP (Ad hoc Network Management Protocol)

ANMP has been developed for managing mobile wireless and ad hoc networks. The protocol uses hierarchical clustering of nodes to reduce the number of messages exchanged between the manager and mobile agents. ANMP is fully compatible with SNMPv3 and used the same PDUs (Protocol Data Units) for data collection.

5.6.6 Interoperability of systems and terminals for broadband multimedia services provided over different network platforms

Interoperability means the capability to provide successful communication between end-user across a mixed environment of different domains, network, facilities, equipment, etc. from different manufacturers and (or) providers. (User Group; User Interoperability Criteria - see bibliography) Interoperability can be qualified at different levels (e.g. protocol interoperability, service interoperability). It is achieved via various types of interworking and interconnection:

- Network interworking - interactions between different network platforms, end-systems, or parts thereof, with the aim of providing an end-to-end communication for a specific service.

- Service interworking.
- Terminal and peripheral interworking and interconnection.

For multimedia services, interoperability can be considered in terms of reliable end-to-end multimedia operation across a number of different networks. In other works, two terminal equipment should be able to operate in a compatible manner by the use of compatible interfaces, coding schemes, information media and control protocols after connection is established. In general, the communication between two terminal equipment can be realized by the following procedure:

- The connection of terminals through the network.
- The identification and selection of the partner's terminal.
- The connection establishment and selection of the parameters and know about the performance of the network.
- The selection of the QoS (network QoS and terminal QoS).
- Transfer the information data, coded data, etc.
- Disconnect the network.

To provide the procedure, it is necessary that both terminals can identify the capabilities of the terminal in the other end to provide a specific service/application. It is thus necessary to register the defined service/application capabilities, possibly through some kind of an information database.

ITU-T Recommendation SG 16 and ITU-T Recommendation MEDIACOM 2004

An audio-visual system Recommendations over different networks has been specified within the ITU-SG 16 (H.3xx set of Recommendations). ITU-T Recommendation SG 16 has undertaken also the task of managing the process of harmonization of new multimedia systems and services and ensuring their end-to-end interoperability within the ITU-T special project MEDIACOM 2004.

ETSI PTCC (Protocol and testing Competence Centre)

The ETSI Protocol and Testing Competence Centre is a unique resource available to ETSI Technical Bodies for the application of leading-edge specification, validation and testing techniques in ETSI deliverables. The task of the PTCC is to help the ETSI membership produce the very best standards and products possible.

Major PTCC achievements include assisting ETSI Technical Bodies in the development of test and/or protocol specifications for:

- Mobile terminals for GSM, 3G and TETRA.
- Wireless LANs, including HiperLan/2.
- Cordless phones DECT.
- VoIP-related protocols (e.g. SIP, H.323).
- Intelligent Networks (INAP).
- APIs for Open Service Architecture (OSA).
- ISDN, Broadband-ISDN, VB5 etc.

The PTCC works closely with the **ETSI Plugtest Service**. Interoperability events organized by this service are a pragmatic and cost-effective means to validate standards and products.

5.6.7 Multimedia mobility

5.6.7.1 Nomadism, mobile communication and ad-hoc networking

Basically, there exist three scenarios for supporting mobility in the user's access to the communication service (Adam Wolisz: "Mobility in Multimedia Communication" - see bibliography).

In the first one, frequently referred to as "nomadic", the travelling customer is interested in mobile access to communication services in different locations, but he is not communicating while moving. In fact two cases of this scenario might be identified:

- Nomadic terminal case: The moving customer uses always his own terminal (i.e. a telephone, workstation, laptop, PDA, etc.). A terminal can be moved from one location to another, while maintaining communication (with or without data movement). The major challenge in this case is the reachability under the original address in the actual, temporary environment (frequently referred to as roaming) as well as security considerations (both for access to the data and the security of communication). While a nomadic terminal case is typically associated with wireless access, wired mobility, i.e. the ability to plug in a terminal at different locations, is also considered.

- Personal nomadicity case: The customer is using each time a different, just locally available at the visited site, terminal. In this case the main issues are: support of a single, universal address assigned to the end-user rather than to his terminal. It is in fact the ability of end users to originate and receive calls and access the subscribed communication services on any terminal in any location. Thus, the network has to be able to identify end users as they move. Personal nomadicity case is based on the use of a unique personal identity (i.e. "personal number").

The second scenario, frequently called mobile communication, supports the use of customer services while moving. As for this scenario some kind of wireless communication is needed, at least on the last hop between the user's terminal and an access point to the fixed communication infrastructure, which is usually referred to as base station. Essential challenges in this case are roaming (similar to the first scenario), but also handover. In addition to that, the specific features of wireless communication impose new challenges on the QoS.

The third scenario, frequently referred to as ad-hoc networking, support the use of some customer services for a closed group of customers who happened to be from some reason on a relatively closed area and need to communicate (emergency services are one example, but other ones like business or project meetings are possible, too). For this case, the usage of wireless communication technology is typical again. The main challenge is to establish communication and service support for a group with non-predictable size, probably with dynamic change both of the number of members as well as their actual position. Therefore there is a need for continuous adjustment of the topology of the network and routing. The issues of QoS assurance arise as well.

5.6.7.2 GMM (Global Multimedia Mobility)

The Global Multimedia Concept was defined was defined by ETSI in the Global Multimedia Report [146],[147]. It denotes the mobility aspects resulting from the convergence of telecommunications, information technology and entertainment services as envisaged by EII/GII. A basic assumption of the GMM concept is that, in the future, terminals should be able to be connected to several types of access network. The choice of access network will be made dynamically and will depend on a variety of factors such as the application requested by the user, the user's subscription, and the access networks available to the user. A variety of access networks can be identified and these include UMTS, DECT access, satellite (Satellite-Personnel Communications System), GSM-BSS (Global System for Mobile communications - Base Station System) and fixed access. The GMM report indicates that the dynamic use of multiple access networks will enable high bit-rate services to be introduced gradually according to market demand. It identifies several core networks and a variety of applications, which reside outside the core network and which are normally transparent to both the access network and the core network.

In the original GMM model the following have been identified:

- independence within the terminal equipment domain (both for fixed and mobile users);

- a multiple access domain (which accommodates both for public and private network solutions);

- a core domain containing different kinds of services networks (including network intelligence needed to support specific service offerings in the core).

The refined GMM model has introduced a service-provisioning domain (in the original GMM model implicitly embedded in the terminal domain) and has opened up the model to visualize a variety of traffic cases (see figure 11). The distant end may consist of services, as shown in the figure, but it may equally be another user and the result may be a "conventional" communication, such as a telephone call. In such a case the right-most domain in the figure (shown as a Service Provisioning Domain) would be replaced by another Terminal equipment (User) Domain, as at the left of the figure. On the other hand, origination of a communication may also be from the right hand side of the figure. In fact, it could be from almost any part of the model where the necessary "intelligence" resides. Base don this model, various other scenarios can be constructed, such as communication between two Service Provider Domains (where no conventional user may be involved).

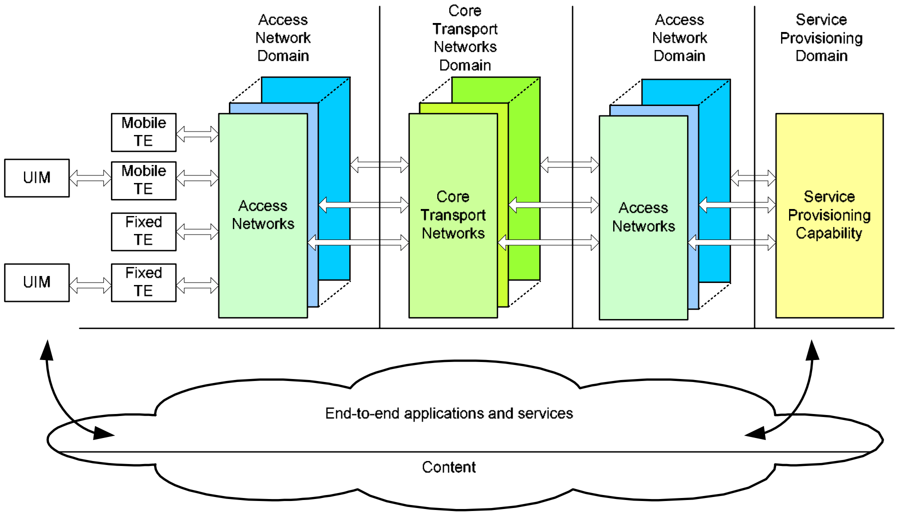

Figure 11: Refined GMM model

The refined GMM model [146] underlines also the necessity of clear separation of the application and services elements from the basic connectivity components. The connectivity part - the "network" in traditional terms - becomes transparent to the different types of information carried.

5.6.7.3 ETSI EP TIPHON mobility service

Mobility in TIPHON compliant systems is based on the assumption that the five TIPHON scenarios [148] include end-to-end voice services when the user is mobile. Different aspects of mobility are distinguished as follows [149]:

- **User mobility** - is the ability of a user to connect to, or use, different terminals or terminal types for the purpose of communication.
- **Terminal mobility -** is the ability for a terminal to change physical location, and still be able to communicate. There are two forms of terminal mobility:
  - *Discrete terminal mobility* (roaming)*:* the ability of a terminal to make discrete changes of physical location. I.e. to change location while no media streams are active.
  - *Continuous terminal mobility* (handover)*:* the ability of a terminal to change physical location, while media streams are active. This can be seamless, no data loss during the change of physical location, or not seamless, in which case some media stream data is lost.
- **Service mobility -** is the ability for a user to obtain a particular service independently of user mobility (i.e. the terminal that they are using) and terminal mobility (i.e. their change of location).

The VHE (Virtual Home Environment) is an extension of service mobility in the sense that it relates to a service package rather than a single service.

Mobility in the context of the five TIPHON scenarios addresses service mobility and end-to-end voice services, when the user is mobile. Within the content of a particular service, a user might have to access, authenticate with or register with, a server in the network that provides the service. For some services this might involve only a server in the home network, or only a server in the visited network, whereas for other services interworking with servers in a visited network might also be needed.

5.6.7.4 H.323 mobility

ITU-T Recommendation SG 16 is currently working on the Draft Recommendation H.510 - "Mobility for multimedia systems and services" (Draft Recommendation H.510 - Mobility for H.323 multimedia systems and terminals - see bibliography). The Draft Recommendation introduces new functionality to H.323 that enables mobility in H.323 systems. Three aspects of mobility are distinguished:

- Terminal mobility: The ability of a terminal to change location (i.e. network point of attachment and H.323 point of attachment) and still be able to communicate.
    - Discrete terminal mobility (terminal roaming): the ability of a terminal to make discrete changes of location, i.e. to change location while no media streams are active.
    - Continuous terminal mobility (handover): the ability of a terminal to change location while media streams are active. Handover is further called seamless when the terminal location change does not result in delay or loss of data that would be perceived by the user as degradation of quality of service (note that seamless handovers may depend on many factors, including service type and service presentation robustness against data loss at the terminal).
- User Mobility (Personal Mobility): The ability of a user to maintain the same user identity irrespective of the terminal used and its network point of attachment. Terminal used may be of different types:
    - Discrete user mobility (user roaming): the ability of a user to change location or terminals while no media streams are active.
    - Continuous user mobility (session mobility): the ability of a user to change location or terminals while media streams are active (a similar feature is provided in the SCN by the supplementary service Terminal Portability).
- Service mobility: The ability of a user to use the particular (subscribed) service irrespective of the location of the user and the terminal that is used for that purpose.

The Draft Recommendation deals with mobility aspects for H.323 systems above the transport layer. H.510 applies new functions defined in support of mobility management to H.323 compliant systems.

ITU-T Recommendation SG 16 is currently working on other new recommendations:

- H.500 - General Mobility Management for Multimedia Systems and Terminals.
- H.520 - Mobile Presence and IM for Multimedia Systems and Terminals.

5.6.7.5 3GPP

3GPP has specified mobility for UMTS OSA APIs [150]. The document specifies the Mobility SCF (Service Capability Feature) aspects of the interface.

5.6.7.6 IETF

The Mobile IP Working Group has developed routing support to permit IP nodes (hosts and routers) using either IPv4 or IPv6 to seamlessly "roam" among IP subnetworks and media types. The Mobile IP method supports transparency above the IP layer, including the maintenance of active TCP connections and UDP port bindings. Where this level of transparency is not required, solutions such as DHCP (Dynamic Host Configuration Protocol) and dynamic DNS updates may be adequate and techniques such as Mobile IP not needed. Recently, the Internet Draft "Mobility Support in IPv6" has been published (Mobility Support in Ipv6 - see bibliography). The protocol defined in the draft document, known as Mobile IPv6, allows a mobile node to move from one link to another without changing the mobile node's "home address". Packets may be routed to the mobile node using this address regardless of the mobile node's current point of attachment to the Internet. The mobile node may also continue to communicate with other nodes (stationary or mobile) after moving to a new link. The movement of a mobile node away from its home link is thus transparent to transport and higher-layer protocols and applications.

Recently, the Network Mobility Working Group has been created within the Internet Area of the IETF. This WG is concerned with managing the mobility of an entire network, which changes, as a unit, its point of attachment to the Internet and thus its reachability in the topology. The mobile network includes one or more MRs (Mobile Routers) which connect it to the global Internet. A mobile network is assumed to be a leaf network, i.e. it will not carry transit traffic. However, it could be multihomed, either with a single MR that has multiple attachments to the Internet, or by using multiple MRs that attach the mobile network to the Internet.

The WG will take a stepwise approach by standardizing some basic support mechanisms based on the bidirectional tunneling approach, and at the same time study the possible approaches and issues with providing more optimal routing than can be had with (potentially nested) tunneling.

The WG will work on:

- threat analysis and security solution for the basic problem (tunneling between Home Agent - HA and MR);
- solution to the basic problem for both IPv4 and IPv6, which will allow all nodes in the mobile network to be reachable via permanent IP addresses, as well as maintain ongoing sessions as the MR changes its point of attachment within the topology. The WG will investigate reusing the existing Mobile IPv6 mechanisms for the tunnel management, or extend it if deemed necessary.

6 Current situation in the standardization bodies

In this clause, an overview of the known key standardization bodies in the domain of technologies for broadband multimedia services and NGN is provided. The clause includes also related bodies, i.e. those that apply generic standards for technologies for broadband multimedia services and NGN to particular applications and/or domains.

6.1 ETSI (European Telecommunications Standards Institute)

6.1.1 Key ETSI TCs and Projects

ETSI TC SPAN (Services and Protocol for Advanced Networks)

ETSI TC SPAN is ETSI's core competence centre for fixed networks standardization including IP based networks, especially for the development of signalling protocols. It is responsible for all aspects of standardization for present and future converged networks including mobility aspects within fixed networks, using existing and emerging technologies, in line with, and driven by, the commercial objectives of the ETSI membership.

The activities are organized in working groups as follows:

- SPAN11 (Services, applications, numbering, naming and addressing) - the group refers to the evolution of services and the specification of NAT & firewall resolutions.
- SPAN12 (Application Interfaces for service providers and network operators) - the group refers to the definition of MEGACO packages including the protocols for API interfaces.
- SPAN13 (Peer to peer control protocol) - the group includes the activities on the BICC protocol and DTM.

- SPAN14 (Overall network and system architecture) - the group points to the solutions for mobile/IP/fixed convergence.
- SPAN15 (Network management).

The NGN issues currently under study include requirements, architecture and definition control and user plane protocols, NAR (Numbering, Addressing and Routing), NGN architecture, NGN services, APIs, network management. SPAN is a member of joint SPAN/3GPP/Parlay committee.

ETSI Project TIPHON (Telecommunications and Internet Protocol Harmonization Over Networks)

The objective of EP TIPHON is to support:

- the migration from switched circuit networks to packet-based networks with an architecture that can serve in both;
- the market for real-time telecommunication services between users, including voice and multi-media communication (like telephony, multi-media conferencing, instant messaging and e-commerce) over multiple network technologies (this is included within NGN).

EP TIPHON addresses:

- service-level inter-working between traditional SCNs, especially those served by PSTN, ISDN or GSM networks, and the emerging NGNs;
- specification and analysis of inter-working scenarios across multiple service, administrative and network technology domains;
- definition of a generic means of creating services that is independent of any specific underlying network technology - irrespective of whether it is switched circuit or packet based.

In doing so, EP TIPHON is considering in the area of its responsibility a broad and diverse set of technical, commercial and legal requirements including security, lawful intercept, emergency communications, quality of service, numbering/naming, billing, mobility and the very approach to communications standardization itself.

EP TIPHON co-operates closely with relevant groupings in ITU-T and IETF and with relevant Fora, e.g. ATM Forum, TeleManagement Forum.

ETSI TC STQ (Speech processing, Transmission and Quality Aspects)

The objective of STQ is to ensure the co-ordination, production (where appropriate) and maintenance of end-to-end speech quality related deliverables, for the timely and economic development of equipment for use with existing and future fixed/mobile network telecommunications service offerings from network operators.

The NGN issues are related mainly to end-to-end QoS.

STQ has also a working group Aurora that meets independently to develop and standardize an algorithm for DSR (Distributed Speech Recognition). The aim is to enable speech to be sent over low quality links such as mobile radio and converted to text for interacting with automated systems.

ETSI TC AT (Access and Terminals)

AT is the "home" for terminal matters within ETSI, established on the basis of a technical area and on the global market sector of telecommunications terminals. AT is organized around a set of ETSI work items addressing specific technology and regulatory areas.

AT centres its NGN activities especially on the field of interaction between terminals and NGN architecture. WG NGN@Home is the WG of TC AT responsible for deliverables relating to Next Generation Networks in the Home environment. To differentiate between the home access and the network access, this WG covers the Home Access Network, which for this purpose may be regarded as the network, which enables the various communications devices within the home to communicate with each other and with the outside world. The WG refers to existing Network Access technologies already defined or to be defined by ETSI and ITU-T documents. This group covers the characteristics and functionality of devices on the Internet that may use the various access networks to transport information across the Home Access Network to the end devices on the Home Local Network.

ETSI TC HF (Human Factors)

ETSI HF is the committee responsible for standards and guidelines dealing with ease of use and accessibility of telecommunication equipment and services, including the requirements of older and disabled people. Human Factors is the scientific application of knowledge about the capacities and limitations of users with the aim of making products, systems, services and environments safe, efficient and easy to use.

The NGN related issues include UCI (Universal Communications Identification), the positioning of letters on standard mobile 12-key keypads and the use of language-specific characters, human factors work in call centres, and multimodal interaction, communication and navigation, access problems of young people (the under 12s).

ETSI SAGE (Security Algorithms Group of Experts)

SAGE is responsible for creating ETSI reports (containing confidential specifications) in the area of cryptographic algorithms and protocols specific to fraud prevention/unauthorized access to public/private telecommunications networks and user data privacy.

ETSI TC TM (Transmission and Multiplexing)

TM works in close co-operation with CENELEC and regularly provides specifications for optical cables and components to be used in various telecommunications environments. This includes WDM (Wavelength Division Multiplexing) components which allow multiple transmission systems to co-exist on a single fibre pair (e.g. in a Passive Optical Network - PON). TM has also begun to study the subject of "All Optical Networks", which considers a future where, not only will cables carry optical signals but the switching will also be done optically (as opposed to electronically).

TM is developing a generic standard covering the functionality of equipment in the transport network, which caters for both PDH and SDH networks (Plesiochronous Digital Hierarchy and Synchronous Digital Hierarchy). Similar work has commenced on a standard for ATM (Asynchronous Transfer Mode) transport network equipment.

NGN related issues include definition of the architecture for the access network and collaboration in the standardization of network interfaces (e.g. for Integrated Subscribers Digital Network (ISDN) and Broadband ISDN (BISDN)). TM is producing specifications for DSL systems for metallic cables. These include ADSL, HDSL, SDSL and VDSL.

6.1.2 Related ETSI TCs and Projects

ETSI Project BRAN (Broadband Radio Access Networks)

The project prepares standards for equipment providing broadband (25 Mbit/s or more) wireless access to wire-based networks in both private and public environments, operating in either licensed or license exempt spectrum. These systems address both business and residential applications.

NGN and broadband multimedia issues are covered by standards for broadband radio access in various environments.

ETSI TC SES (Satellite Earths Stations and Systems)

ETSI SES is responsible for all types of satellite communication services (including mobile and broadcasting) and for all types of earth station equipment (especially the radio frequency interfaces and network and/or user interfaces). This includes definition of satellite system architecture supporting broadband services, service requirements and descriptions for broadband communication systems, definition of network architectures and interface protocols leading to air interface standards, interworking standards and other user terminal specifications.

NGN and broadband multimedia issues are covered by standards for broadband satellite multimedia (Broadband Satellite Multimedia WG formed in June 2002).

ETSI TC ESI (Electronic Signatures and Infrastructure)

TC ESI is the lead body within ETSI in relation to Electronic Signatures and Infrastructures. It addresses some basic needs of secure electronic commerce and of secure electronic document exchange in general by providing specifications for a selected set of technical items that have been found both necessary and sufficient to meet minimum interoperability requirements. Examples of business transactions based on electronic signatures and public key certificates are purchase requisitions, contracts and invoice applications.

6.2 ITU-T

6.2.1 Key ITU-T SGs and ITU-T Special Projects

ITU-T Recommendation SG 16

SG 16 is responsible for studies relating to multimedia service definition and multimedia systems, including the associated terminals, modems, protocols and signal processing. It is the lead Study Group on multimedia services, systems and terminals as well as the lead Study Group on e-business and e-commerce.

Several study questions under the responsibility of SG 16 are directly related to broadband multimedia services and the relevant technologies, e.g. multimedia architecture, multimedia applications and services, interoperability of multimedia systems and services, media coding, QoS and end-to-end performance in multimedia systems, security of multimedia systems and services, accessibility to multimedia systems and services, multimedia systems, terminals and data conferencing, multimedia over packet networks using H.323 systems, voice coding techniques, etc.

ITU-T Recommendation SG 16 has established the ITU-T project - MEDIACOM 2004.

ITU-T Project MEDIACOM 2004

The objective of the Mediacom 2004 Project (MEDIACOM 2004, A Framework for Multimedia Standardization, Project Description - see bibliography) is to establish a framework for multimedia standardization for use both inside and external to the ITU. This framework should support the harmonized and coordinated development of global multimedia communication standards across all ITU-T and ITU-R Study Groups, and in close cooperation with other regional and international SDOs (Standards Development Organizations). The scope of Mediacom 2004 includes the following applications:

- end-to-end multimedia systems and services over all network types including the Internet: videophone/videoconference, multipoint/multicast multimedia systems, multimedia on demand, electronic commerce, distance learning, tele-medicine, interactive TV services, web-casting, MBone, including their distribution within the home environment, etc.;
- end-to-end multimedia systems and services over wireless access systems, e.g. using Radio Frequency or Infrared (IMT-2000, Wireless Application Protocol Forum, Bluetooth, HomeRF, IrDA, etc.);
- security system for using multimedia systems (watermark in the video contents, individual authentication, etc.);
- multimedia broadcasting systems that interactively handle audio and video;
- the extension of e-mail and the WWW for the transmission of multimedia documents.

Within the framework of the Mediacom 2004 Project, SG 16 is responsible for the following technical issues:

- developing a framework for multimedia services and systems that must be, as far as possible, independent from the underlying infrastructure;
- developing appropriate services and systems related standards for applications in multimedia communications;
- ensuring that end-to-end interoperability is accommodated either by full compatibility between systems or by specification of the appropriate gateways.

ITU-T Recommendation SG 13

SG 13 is responsible for studies relating to internetworking of heterogeneous networks encompassing multiple domains, multiple protocols and innovative technologies with a goal to deliver high-quality, reliable networking. Specific aspects are architecture, interworking and adaptation, end-to-end considerations, routing and requirements for transport. It is the lead Study Group for IP-related matters, B-ISDN, Global Information Infrastructure and satellite matters.

The question currently under study related to NGN includes especially architectural and interworking issues between NGN and IP networks.

ITU-T IP Project

The scope of the IP project (ITU-T IP Project Description - Version 7 - see bibliography) involves the following twelve work areas:

Area 1: Integrated architecture.

Area 2: Impact to telecommunications access infrastructures of access to IP applications.

Area 3: Interworking between IP based network and switched-circuit networks, including wireless based networks.

Area 4: Multimedia applications over IP.

Area 5: Numbering and addressing.

Area 6: Transport for IP-structured signals.

Area 7 Signalling support, IN and routing for services on IP-based networks.

Area 8: Performance.

Area 9: Integrated management of telecom and IP-based networks.

Area 10: Security aspects.

Area 11: Network capabilities including requirements for resource management.

Area 12: OAM (Operations and Maintenance) for IP.

Area 13: Utilization of IPv6 in telecommunication networks.

ITU-T Project NGN 2004

During its January 2002 meeting, SG 13 decided to undertake the preparation of a new ITU-T Project entitled "NGN 2004 Project" (NGN 2004 Project Description - see bibliography). At the November 2002 SG 13 meeting, a preliminary description of the Project was achieved and endorsed by SG 13 with the goal to launch the project. The role of the NGN 2004 Project is to organize and to coordinate ITU-T activities on Next Generation Networks. Its target is to produce a first set of Recommendations on NGN by the end of this study period, i.e. mid-2004.

Basically, the NGN 2004 Project is seen as a realization of the concepts adopted in the GII (Global Information Infrastructure). The intention of the NGN 2004 Project is to establish implementation guidelines and standards for the realization of Next Generation Networks based on GII concepts. The major task of the NGN 2004 Project will be to describe all elements required for interoperability and network capabilities to support applications globally across Next Generation Networks.

ITU-T Recommendation SG 11

SG 11 is responsible for studies relating to signalling requirements and protocols for Internet Protocol related functions, some mobility related functions, multimedia functions and enhancements to existing Recommendations on access and internetwork signalling protocols of ATM, N-ISDN and PSTN. It is the lead Study Group on intelligent networks.

The current study questions related to NGN include: signalling requirements for signalling support for new, value added, IP based and IN based services, network signalling protocols to support mobility in the fixed network, network signalling requirements for the support of VHE (Virtual Home Environment) in the fixed network, signalling requirements for signalling support for service interworking of both dialup Internet access and voice, data and multimedia communications over IP-based networks, signalling requirements for the support of BICC applications, access and network signalling for advanced narrow-band and broadband services, etc.

ITU-T Recommendation SG 4

SG 4 is Responsible for studies relating to the management of telecommunication services, networks, and equipment using the TMN framework. Additionally responsible for other telecommunication management studies relating to designations, transport-related operations procedures, and test and measurement techniques and instrumentation. It is the lead Study Group on TMN.

The NGN related issues are covered by the definition of the framework for unified management of integrated circuit-switched and packet-based networks (with an initial emphasis on IP-based networks).

ITU-T Recommendation SG 15

SG 15 is the focal point in the ITU-T for studies on optical and other transport networks. It is also the lead SG on access network transport and on optical technology.

Recently, the ANT (Access Network Transport) Project (Access Network Transport - ITU-T Recommendation SG 15 Project - see bibliography) has been established by SG 15. Access Network Transport is a project dealing with studies and Recommendations on the different types of access networks.

6.2.2 Related ITU-T SGs and ITU-T Special Projects

ITU-T Recommendation SG 9

SG 9 prepares and maintains Recommendations on the use of cable and hybrid networks, primarily designed for television and sound programme delivery to the home, as integrated broadband networks to also carry voice or other time critical services, video on demand, interactive services, etc., as well as the Recommendations on the use of telecommunication systems for contribution, primary distribution and secondary distribution of television, sound programmes and similar data services. It is also the lead Study Group on integrated broadband cable and television networks. The NGN related issues include especially IP Cablecom applications, video quality measurement techniques for digital cable television, harmonization of procedural content formats for interactive TV applications, transport of D-cinema applications that employ MPEG-2 encoded HDTV signals.

ITU-T IMT-2000

IMT-2000 represents the global standard for meeting the emerging needs of mobile telecommunications in the 21st century whereby mobile telecommunications subscribers will be able to access voice, data, Internet, and multimedia services at any time and at any place. It includes production of standards that will not only ensure seamless global mobility and service delivery across IMT-2000 family member networks, but that will also integrate the wireline and wireless networks in order to provide telecommunication and information services transparently to the users.

The work on the network aspects of mobility issue is carried out in the SSG (Special Study Group) "IMT2000 and Beyond". It includes wireless Internet, convergence of mobile and fixed networks, mobility management, mobile multimedia functions, internetworking, interoperability and enhancements to existing ITU-T Recommendations on IMT-2000.

ITU-T Special Project JVT (Joint Video Team)

JVT is the joint project of ITU-T Recommendation Q.6/SG 16 ("VCEG") and ISO/IEC JTC1/SC29/WG11 ("MPEG"). The goal of the joint project is to develop a new video coding standard satisfying the high-level requirements set. The work will have as starting point the current draft ITU-T Recommendation H.26L developed by VCEG that has proved to provide the most advanced results in the tests carried out by MPEG in June 2001.

The scope of the joint project will be not only the development of a new video coding standard, but also the assessment of its performance at the completion of the work using formal subjective testing procedures.

The intent is that the ITU-T Recommendation and ISO/IEC International Standard be technically aligned, fully interoperable with each other for all of the video codec's conformance points specified during the term of the joint work, and offer the best possible technical performance under the practical constraints of being implementable on various platforms and for various applications enabled by the relevant ITU-T Recommendations and ISO/IEC International Standards.

ITU-T Special Project IP Cablecom

IP Cablecom is the ITU-T Special Project on time-critical interactive services over cable television network using IP-protocol, in particular Voice and Video over IP.

ITU-R

ITU-R broadband multimedia related activities as mentioned in clause 5.6.2.1.6.

6.3 ISO (International Organization for Standardization)

Within the ISO, the JTC1 (Joint Technical Committee) is responsible for developing standards for information technology.

The NGN-related issues are covering mainly by JTC/SC 6 (Telecommunications and information exchange between systems), JTC1/SC 27 (IT security techniques), and JTC1/29 (Coding of audio, picture, multimedia and hypermedia information).

6.4 ANSI (American National Standards Institute)

Within ANSI, it is Standards Committee T1 who develops American National Standards, technical reports and technical requirements for telecommunications services, network interconnection, interoperability, and performance. Committee T1 provides technical input to the United States Department of State supporting U.S. participation in international standards bodies. Specifically, T1 focuses on those functions and characteristics associated with the interconnection and interoperability of telecommunications networks at interfaces with end-user systems, carriers, and information and enhanced service providers. These include switching, signalling, transmission, performance, operation, administration and maintenance aspects. Committee T1 is also concerned with procedural matters at points of interconnection, such as maintenance and provisioning methods and documentation, for which standardization would benefit the telecommunications industry. Committee T1 is a founding member of the GSC (Global Standards Collaboration) group of regional standards development organizations and works closely with the U.S. FCC (Federal Communications Commission) on network reliability issues.

The NGN-related issues are covering by the following subcommittees:

- **T1A1 (Performance and Signal Processing):** QoS, IP network performance/reliability, coding for user plane traffic.
- **T1E1 (Network Interfaces, Power and Protection):** UNI interface standards for DSL, optical and other electrical access networks.
- **T1M1 (Internetwork Operations, Administration, Maintenance and Provisioning):** network management, XML/tML framework.
- **T1P1 (Wireless/Mobile Services and Systems):** mobility issues.
- **T1S1 (Services, Architectures, and Signalling):** BICC, SIGTRAN, signalling architecture.
- **T1X1 (Digital Hierarchy and Synchronization):** optical transmission.

6.5 IETF

The actual technical work of the IETF is done in its working groups, which are organized by topic into several areas (e.g. routing, transport, security, etc.). Much of the work is handled via mailing lists. The IETF holds meetings three times per year.

6.5.1 Key IETF Technical Areas and Working Groups

Key technical areas and working groups involved in multimedia and NGN matters are:

- Transport Area (e.g. avt - Audio/Video Transport, enum - Telephone Number Mapping, megaco - Media Gateway Control, midcom - Middlebox Communication, mmusic - Multiparty Multimedia Session Control, nsis - Next Steps in Signalling, sigtran - Signalling Transport, sip - Session Initiation Protocol, sipping - Session Initiation Proposal Investigation, speechsc - Speech Services Control, spirits - Service in the PSTN/IN Requesting InTernet Service).
- Sub-IP Area (e.g. mpls - Multiprotocol Label Switching).

It should be noted that IETF is essentially a "protocol factory" and so is normally not involved in architecture issues.

6.5.2 Related IETF Technical Areas and Working Groups

Related technical areas and working groups involved in multimedia NGN matters are:

- Applications Area (calsch - Calendaring and Scheduling, cdi - Content Distribution Internetworking, ediint - Electronic Data Interchange-Internet Integration, geopriv - Geographic Location/Privacy, impp - Instant Messaging and Presence Protocol, simple - SIP for Instant Messaging and Presence Leveraging Extensions, vpim - Voice Profile for Internet Mail, xmpp - Extensible Messaging and Presence Protocol, etc.).

- Security area (idwg - Intrusion Detection Exchange Format, inch - Extended Incident Handling, ipsec - IP Security Protocol, ipsec - IP Security Policy, krb-wg - Kerberos WG, msec - Multicast Security, pkix - Public-Key Infrastructure (X.509), smime - S/MIME Mail Security, syslog - Security Issues in Network Event Logging, tls - Transport Layer Security, xmldsig - XML Digital Signatures, etc.).

- Internet Area (ipcdn - IP over Cable Data Network, ipv6 - IP Version 6 Working Group, nemo - Network Mobility, etc.).

- Operations and Management Area (aaa - Authentication, Authorization and Accounting, snmpconf - Configuration Management with SNMP, snmpv3 - SNMP Version 3, etc.).

6.6 Other bodies

In this clause, some other bodies involved in the standardization process in the domain of technologies for broadband multimedia services and NGN will be overviewed.

6.6.1 ECMA (Europe-based Association for Standardizing Information and Communication Systems)

TC32 is a Technical Committee of the Europe-based Association for Standardizing Information and Communication Systems (ECMA). Under a co-operation agreement between ECMA and ETSI, by which the two organizations agree to share responsibility for standardization in the field of private/corporate telecommunications networks, TC32 acts as a Technical Committee of ETSI.

Because CNs (Corporate Networks), unlike public networks, must operate homogeneously across national boundaries standards should be applicable worldwide. Therefore standards created within TC32 are fed into the international standardization organizations (Joint Technical Committee 1 (JTC1) of the ISO and the IEC).

Key technologies standardized by ECMA TC32 include:

- PISNs (Private Integrated Switched Networks), including QSIG (Signalling at the Q Reference Point).

- CSTA (Computer-supported Telecommunications Applications).

- Broadband PISNs.

- PISN - IP interoperability.

Next Generation Networks for enterprises will definitely use the IP as the basis for signalling and media transport. To reflect this development a new Task Group (TC32-TG17) was established in 1999, starting with interworking of call signalling and call control services for voice communication between PISNs and IP networks. Current projects cover specifications on QSIG/H.323 interworking including mapping and tunnelling of QSIG messages, transport of QSIG over TCP/IP and QSIG to SIP message mapping. As part of the activities in TC32-TG17, members have produced an Internet-Draft on interworking between QSIG and SIP for submission to the IETF. Further work items are going to address the specification of SIP-based call control services taking into account enterprise needs.

6.6.2 DSL Forum

DSL Forum is a consortium of more than 400 leading industry players covering telecommunications, equipment, computing, networking and service provider companies. Established in 1994, the Forum continues its drive for a mass market for DSL, to deliver the benefits of this technology to end users around the world over existing copper telephone wire infrastructures.

Best practices for auto-configuration, flow through provisioning and a range of other key facilitators of scaleable, global, mass-market deployment of DSL technology are fast-tracked by DSL Forum through its Technical Committee and Marketing Committee working groups. This work takes place at quarterly, week-long meetings and through continuous working group progress programmes with formal technical reports developed from contributions and "Working Texts".

NGN related issues include especially application of NGN concepts to a DSL network and voice over multi-service data network.

6.6.3 DVB (Digital Video Broadcasting)

The Digital Video Broadcasting Project is an industry-led consortium of over 300 broadcasters, manufacturers, network operators, software developers, regulatory bodies and others in over 35 countries committed to designing global standards for the global delivery of digital television and data services.

The scope of the DVB has been widened to build a content environment that combines the stability and interoperability of the world of broadcast with the vigour, innovation and multiplicity of services of the world of the Internet. The core of DVB's new mission is to provide the tools and mechanisms to facilitate interoperability and interworking between different networks, devices and systems to allow content and content based services to be passed through the value chain to the consumer.

DVB systems are developed through consensus in the working groups of the Technical Module. Members of the groups are drawn from the general assembly of the project. Once standards have been published, through ETSI, they are available at a nominal cost for anyone, worldwide. Open standards free manufacturers to implement innovative and value added services. It does not matter where DVB technology is developed. It is available worldwide.

NGN related issues include mainly a transport of multimedia content via IP networks, MHM (Multimedia Home Platform) and mobile-broadcast convergence architectures.

6.6.4 IMTC (International Multimedia Telecommunications Consortium)

IMTC is an international community of companies working together to facilitate the availability of real-time, rich-media communications between people in multiple locations around the world. Rich Media refers to converged communications sessions that incorporate voice and one-way (or two-way) data and one-way (or two-way) video.

The IMTC Focuses on:

- promoting standards that enable real-time, rich-media communications.
- identifying obstacles to ubiquitous utilization of multimedia products and services.
- developing and submitting technology interoperability recommendations to official standards bodies -such as the ETSI, IETF, 3GPP, ISO, and ITU-T.
- initiating scheduled interoperability test sessions between suppliers of rich-media products and services.

6.6.5 IPCC (International Packet Communications Consortium)

The International Packet Communications Consortium evolved from the ISC (International Softswitch Consortium, the industry's most longstanding advocate advancing the maturation of packet-based network technologies and markets. The IPCC embodies the industry's primary mission: To develop the market for all products, services, applications and solutions utilizing packet-based voice, data and video communications technologies available today, regardless of transport medium - wireless, copper, broadband, fibre optics and more.

There are currently eight active working groups in the IPCC:

Applications WG: The purpose of this working group is to define interaction requirements that are necessary for application servers to provided enhanced services. Providing interaction requirements will allow faster introduction of enhanced services that combine telephony, Internet, messaging and other user content. It will also promote interoperability that can easily be tested in the IPCC Interoperability Test Lab.

Billing WG: The purpose of the new Working Group is to determine a standard billing interface method of the softswitch based network.

Interoperability WG: The aim is to define the goals, requirements, and technical criteria necessary to develop an industry standard end to end VOIP interoperability test suite, to provide input and guidance to test writers and implementers and to plan IPCC interoperability events.

Legal Intercept WG: The objective is to produce a document(s) describing the requirements for Legal Intercept. Also a document diagramming how legal intercept works in the IPCC architecture. This group will function as a forum for obtaining and distributing information to members regarding interfacing with legal and regulatory agencies.

Marketing WG: The group seeks to clearly state IPCC's missions and goals and also generate a significant increase in the acceptance and implementation of the IPCC's proposed architecture and protocols.

Network Boundary Functionality WG: The purpose is to document and review requirements from the carriers and the Enterprise/SOHO with respect to VoIP and "filtering" boxes at the edge of the networks.

SIP WG: The primary purpose of this working group is to address the signalling of packet-based IP Networks and the interworking of packet-based IP network signalling with the PSTN signalling. For interworking with PSTN, IP networks will need to transport signalling such as Q.931 and SS7 ISUP messages between IP nodes such as SoftSwitches and SIP Proxy servers. The interconnection of packet networks through secure Proxy servers is also necessary for the development of SoftSwitch based networks that exchange packet-based traffic.

Wireless WG: The purpose of this group is to evangelize the Softswitch model for use in 2G and 3G wireless core networks where wireless Softswitchs (MSC Server), Signalling Gateways and Media Gateways will eventually replace legacy Mobile Switching Centres (MSCs) in cellular core networks. Additional activities include interoperability tests in the IPCC Interoperability Test Lab. Other possible activities include creation of Test Scripts, interface requirements and gap analysis of various standards (UMTS/GSM/CDMA 3GPP/3GPP2 etc.) as it applies to softswitch architecture.

6.6.6 JAIN

The JAIN APIs are a set of Java technology based APIs which enable the rapid development of Next Generation telecom products and services on the Java platform. The JAIN APIs bring service portability, convergence, and secure network access to telephony and data networks. By providing a new level of abstraction and associated Java interfaces for service creation across PSTN, packet (e.g. IP or ATM) and wireless networks, JAIN technology enables the integration of Internet and Intelligent Network protocols. This is referred to as Integrated Networks. Furthermore, by allowing Java applications to have secure access to resources inside the network, the opportunity is created to deliver thousands of services rather than the dozens currently available.

JAIN technology is being specified as a community extension to the Java Platform. Development is being carried out under the terms of Sun's JSPA (Java Specification Participation Agreement, JCP (Java Community Process), and SCSL (Sun's Community Source Code Licensing) terms.

The JAIN initiative consists of two API Specification areas of development:

- Protocol API Specifications specify interfaces to wireline, wireless and IP signalling protocols.
- The Application API Specifications address the APIs required for service creation within a Java framework spanning across all protocols covered by the Protocol API Specifications.

6.6.7 MPLS and Frame Relay Alliance

Founded in March 2000, the MPLS and Frame Relay Alliance is an industry-wide organization of networking and telecommunication companies focused on driving the deployment of multi-vendor MPLS networks and associated applications.

Through the efforts of our three working committees, the Forum encourages: (a) input to the development of standards throughout the various industry standards groups; (b) the creation of Implementation Agreements, based upon appropriate standards, on how to build and deliver MPLS networks and services; (c) the definition of Interoperability test suites and coordination of Interoperability events to demonstrate the readiness of MPLS for network deployments; (d) the creation and delivery of educational programs to educate the industry about MPLS technologies, services and solutions; and (e) building the awareness of MPLS as a technology ready for wide-scale deployment within service provider networks to deliver profitable services to the end-user community.

6.6.8 MSF (Multiservice Switching Forum)

The Multiservice Switching Forum is a global association of service providers and system suppliers committed to developing and promoting open-architecture, multiservice switching systems. The MSF's activities include developing implementation agreements, promoting worldwide compatibility and interoperability, and encouraging input to appropriate national and international standards bodies.

6.6.9 PacketCable

PacketCable is a CableLabs-led initiative aimed at developing interoperable interface specifications for delivering advanced, real-time multimedia services over two-way cable plant. Built on top of the industry's highly successful cable modem infrastructure, PacketCable networks uses Internet protocol technology to enable a wide range of multimedia services, such as IP telephony, multimedia conferencing, interactive gaming, and general multimedia applications. Working with CableLabs member companies and technology suppliers, the PacketCable project addresses issues such as device interoperability and product compliance with the PacketCable specifications.

6.6.10 Parlay Group

The Parlay Group is a multi-vendor consortium formed to develop open, technology-independent APIs that enable the development of applications that operate across multiple, networking-platform environments. Parlay integrates IN services with IT applications via a secure, measured, and billable interface. By releasing developers from underlying code, networks, and environments, Parlay open APIs allow for innovation within the enterprise.

The Parlay Group's mission is to define, establish, and support a common specification for industry-standard APIs, facilitate the production of test suites and reference code in multiple technologies that enable developers to create related products and services that operate across wireless, IP, and public-switched networks, support the creation and implementation of uniform conformance test procedures and processes, which assure that Parlay API implementations are compliant with the specifications.

6.6.11 TIA (Telecommunications Industry Association)

The Telecommunications Industry Association is the leading U.S. non-profit trade association serving the communications and information technology industry, with proven strengths in market development, trade shows, domestic and international advocacy, standards development and enabling e-business. Through its worldwide activities, the association facilitates business development opportunities and a competitive market environment. TIA provides a market-focused forum for its member companies, which manufacture or supply the products and services used in global communications.

The NGN-related issues include mainly interaction between terminals and NGN architecture, lawful interception architecture and network management.

6.6.12 W3C (World Wide Web Consortium)

The World Wide Web Consortium develops interoperable technologies (specifications, guidelines, software, and tools) to lead the Web to its full potential. W3C is a forum for information, commerce, communication, and collective understanding. W3C organizes the work necessary for the development or evolution of a Web technology into Activities. Each Activity has its own structure, but an Activity typically consists of a Working Group, Interest Group, and Coordination Group. Within the framework of an Activity, these groups generally produce Recommendations and other technical reports as well as sample code.

Activities are grouped into 4 domains. The following domains and Activities are currently active within W3C:

- **Architecture:** DOM - Document Object Architecture, Internationalization, URI, Web services, XML.
- **Interaction:** Device Independence, Graphics, HTML, Math, Multimodal Interaction, Style, Synchronized Multimedia, Voice Browser, Xforms.
- **Technology and Society:** Patent Policy, Privacy, Semantic Web, XML Encryption, XML Key Management, XML Signature.
- **WAI (Web Accessibility Initiative):** WAI International Programme Office, WAI Technical Activity.

6.6.13 3GPP (3rd Generation Partnership Project)

The original scope of 3GPP was to produce globally applicable Technical Specifications and Technical Reports for a 3rd Generation Mobile System based on evolved GSM core networks and the radio access technologies that they support (i.e. UTRA (Universal Terrestrial Radio Access) both FDD (Frequency Division Duplex) and TDD (Time Division Duplex) modes). The scope was subsequently amended to include the maintenance and development of the GSM (Global System for Mobile communication) Technical Specifications and Technical Reports including evolved radio access technologies (e.g. GPRS (General Packet Radio Service) and EDGE (Enhanced Data rates for GSM Evolution)).

The 3GPP is currently working on standards covering UMTS and GSM mobile networks and has work items covering the evolution of the core network architecture from circuit switch to packet switch technology with the creation of the IP multi-media domain. There are also enhancements to the service tool kits and radio access capabilities.

NGN issues cover application of NGN concepts to a UMTS based mobile network, service APIs. In particular to the use of NGN technologies within a circuit switched service environment in release 4 and is also as part of the IM (Internet Multimedia) mode operation within release 5.

6.6.14 3GPP2 (Third Generation Partnership Project 2)

3GPP2 is a collaborative third generation (3G) telecommunications specifications-setting project comprising North American and Asian interests developing global specifications for ANSI/TIA/EIA-41 Cellular radiotelecommunication Intersystem Operations network evolution to 3G and global specifications for the radio transmission technologies (RTTs) supported by ANSI/TIA/EIA-41.

3GPP2 was born out of the ITU's International Mobile Telecommunications (IMT-2000) initiative, covering high speed, broadband, and IP-based mobile systems featuring network-to-network interconnection, feature/service transparency, global roaming and seamless services independent of location. IMT-2000 is intended to bring high-quality mobile multimedia telecommunications to a worldwide mass market by achieving the goals of increasing the speed and ease of wireless communications, responding to the problems faced by the increased demand to pass data via telecommunications, and providing "anytime, anywhere" services.

7 Identification of the standards areas concerned and the gaps

7.1 Multimedia services

There is a need to standardize a minimum set of multimedia applications, services and interfaces which will fully meet evolving user needs. So far, there exists the ITU-T Recommendation F.700 [1] (applied to F.700-series of Recommendations) that provides a general methodology for description of services based on a modular approach across different networks. There are still many open issues:

- Definition of a generic broadband multimedia service is still missing.
- Initial road map of multimedia services and applications standardized within ETSI should be provided.
- The requirements for the services and applications provided within the initial road map should be defined (from the user, service provider and network provider point of view).
- Should ETSI endorse the ITU-T Recommendation F.700 series of recommendations [1] and enhance it at IP based services and capabilities, or should ETSI provide for the own general methodology for description of broadband multimedia services?

7.2 Accessibility

Possible areas of further investigation in the domain of xDSL technologies include the issues of interoperability among component manufactures and carriers, technical solutions for elimination of cross-talk interference from nearby wires and unification of power system requirements.

However, there are especially **human aspects** of the accessibility that require standardization. The capabilities to handle different information media and control actions is very varying among users of traditional telecommunication services and will vary a lot with the use of multimedia services. This might be even more evident for ageing population, or disable people. It will be important to take in mind this variety of people capabilities in designing the multimedia services. Thus, the multimedia services should be design in such a way to enable a great number of users may benefit of them.

That is why adequate user control mechanisms over multimedia services and devices should be provided without the necessity to use of all human senses and capabilities. The user control over multimedia services should be possible in alternative ways, assuring that information is provided in alternative media. Thus, the standardized solutions should be identified and designed with the aim to improve the human accessibility of broadband multimedia services.

7.3 Control layer

The ETSI EP TIPHON is developing a generalized communications protocol to support voice services over IP with the emphasis initially on public telephony. This "meta-protocol" protocol is being mapped into actual protocols such as SIP and H.323 with the production of standards that are in effect a combination of profiles (choices of options) and deltas (additions) that define how to use SIP, H.323 and H.248. TIPHON looks at interworking between SIP and H.323 and between these protocols and ISUP to provide inter-operability with circuit switched networks.

The possible areas for standardization within ETSI:

- To develop a generic functional architecture to model the control plane for the NGN taking into account broadband multimedia services (conversational, non-conversational) and all types of public access networks over which the services may be delivered.
- To apply the generic functional architecture to DSL-based access network that will be probably for longer time the preferable access technology for home users and small business users.
- To apply the generic functional architecture to optical-based access networks.
- To standardize signalling protocols for NGN (e.g. based on SIP, H323, etc.).
- To standardize a set of signalling protocol profiles in such a way to be as neutral as possible with respect to the access technology and to ensure the end-to-end support of services across the international network.

7.4 Service layer

Up to now, there is no criteria for selecting the most proper model, or technique for service development. ETSI should provide guidelines for selecting the most appropriate one based on several criteria:

- Network characteristics, e.g. the abstraction of underlying network infrastructure that can be used by application developers to exploit network functionalities; they can represent both functional (e.g. call control) and non-functional (e.g. authentication, logging, etc.) aspects.
- Service characteristics.
- Evaluation of interfaces offered by technologies to developers, e.g. the kind of interface (API, protocol-based interface, scripting language), the level of abstraction, etc.
- Maturity.
- Usability.
- Etc.

ETSI should provide guidelines for selecting the most appropriate model based on different criteria taken into account.

7.5 Issues affecting all layers

7.5.1 Addressing, numbering and naming

Some of the issues which have been identified within the ITU IP Project (ITU-T IP Project Description - Version 7 - see bibliography) include:

- accommodation of E.164 Number Portability;
- allocation of E.164 resources to IP-based user;
- E.164 - DNS interworking (definition of interworking concepts and principles, administrative and operational aspects, the possibility of direct exchange between directories);
- carrier selection/preselection for IP-based users;
- administration of E.164 numbers in DNSs;
- definition of a global administrative framework within ENUM that will secure consistency of mapping between the E.164 dial plan and the parallel DNS structure;
- support of tracking of E.164-name to IP address for wireless and portable terminals.

From a long-term point of view, ITU-IP Project has identified the necessity to specify of a long-term vision of a global entity (e.g. person, legal entity, device equipment) naming scheme for all communication needs.

7.5.2 Media coding

Based on the identification of emerging multimedia services and applications, the corresponding media that need to be encoded/decoded, or otherwise represented, should be identified.

Other areas include security aspects (watermark, privacy) related to media coding, requirements on media coding to provide the same end-to-end quality when using the different networks, requirements on media coding to provide different QoS within the same multimedia service when using the same terminal.

The possible study areas should include also such aspects as the interoperability between existing systems in a view of transcoding-free systems and scenarios and minimization of quality loss in transcoding scenarios.

7.5.3 QoS

Since a multimedia service is a combination or set of combinations of two or more media components (e.g. audio, video, graphics, etc.), it is not appropriate to talk simply about "multimedia QoS". The approach needed is to define for each media component the appropriate QoS classes. Each media component and its QoS classes should be provided within a common generic framework that should be open enough in order to be capable to include new media components that have been not defined yet.

The open issues of standardization within ETSI include:

- Specification of end-to-end quality for multimedia and the impact of different layers on the end-to-end QoS.
- Definition of end-to-end QoS classes for multimedia and recommended QoS traffic parameters, possibly:
  - per media type (e.g. speech, audio, video, still image, data, etc.);
  - at the application level for different applications (IP telephony, audiovisual conferencing, audio streaming, audiographic conferencing, etc.);
- specification of different multimedia service types with the respect to their requirements on synchronizing different service components (i.e. different media);

- specification of how to use lower layer QoS mechanism to achieve upper layer QoS within the network;
- specification of inter-domain lower layer control.

As far as concerns a call set-up quality, the following issues may need to be studied by ETSI:

- definition of delay values in multimedia applications and systems for session and media flow set-up, mid-session changes, session and media flow tear-down;
- definition of reliability and accuracy of session and media flow set-up, mid-session changes, session and media flow tear-down;
- requirements for signalling latency and signalling reliability.

7.5.4 Security

There are a lot of open issues in security for multimedia services. They can be grouped into several domains:

- Security requirements and security services
  - Identification of multimedia applications and services that require security.
  - Identification of security services and mechanisms for the specified multimedia applications and services.
  - Identification of specific security requirements for e-commerce applications.
  - Identification and specification of mechanisms for protection of multimedia content (authorship, IPR, copy-protection).
- Security architecture and infrastructure
  - Security architecture for NGN.
  - Security architecture for distributed multimedia systems and services.
  - Architecture for a security policy, possibly for different grades of a security policy.
  - Security interaction with other factors such as QoS, performance, etc.
- Multimedia communication security
  - Secure signalling for multimedia applications.
  - Secure data transport and the multimedia stream for multimedia applications.
  - Protection of multimedia content in terms of authorship, copyright, copy protection.
- Interdomain security
  - Secure interaction of the distributed multimedia systems across different network domains.
  - Impact of security devices such as firewalls and security gateways on the multimedia security.

7.5.5 Network management

The NGN is based on various forms of combined fixed, mobile, IP, etc. access networks. This fact creates increasing complexities and challenges related to the management of such networks. This also applies to the management of existing and new services, including multimedia ones across different network types. Integrates remote management will be essential to gain the full benefits of integrating various IP and traditional telecom technologies. Based on the facts stated in clause 5.6.5, currently management of IP networks is focused on the use of IETF and some ITU-T management standards while the management of traditional telecom networks is based on the ITU-T TMN Recommendations. It is important to understand the management needs and requirements of both domains in order to develop an integrated network management approach.

The focus should be done on the definition and creation of an integrated NGN management architecture, requirements, network management services and interfaces and protocols.

7.5.6 Interoperability of systems and terminals for broadband multimedia services provided over different network platforms

Interoperability of systems and terminals for broadband multimedia services and applications is an another open domain for standardization. Specifications for multimedia interoperability should be provided:

- In terms of reliable end-to-end multimedia operation across a number of different networks. Interworking between any two networks requires several levels of interworking:
  - conversion between bearer formats;
  - conversion between supported media formats;
  - conversion between multiplexes;
  - signalling interworking.
- In terms of different applications and services (network based or system based), which have to interoperate efficiently and reliably in a multimedia environment.

7.5.7 Mobility

The principal objective should be to introduce mobility to a federation of converged networks, i.e. the possibility to offer roaming (user service delivery) and handover (maintaining a continuous service) via different access technologies and networks in such a way the AAA, security and QoS issues will be considered during handoff; Another problem that should be resolved is the possibility of an end-user to move across different service providers and access domains should be taken into account.

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