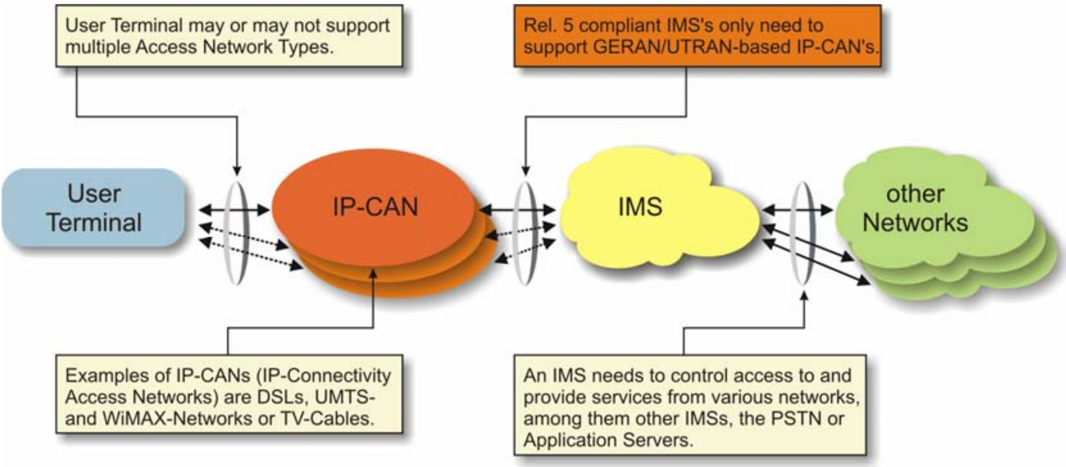
1.1 The IMS within the Overall Network Architecture



The objective of this section is to illustrate the IMS in relationship to other networks and to the user agent.

Key points of this section are:
1. With respect to the IMS, GERAN/UTRAN is reduced to a plain IP-CAN.
2. An important limitation of the IMS with Rel. 5 is the fact that only GERAN/UTRAN is a legitimate IP-CAN's. This is due to security and NAT/NAPT-issues.

- **Relationship to other Networks**
  - ⇒ The IMS takes on the role of a mediator between the user terminal on one side and the services on the other side. Services are provided by "other networks" which include heterogeneous networks like the PSTN, other IMS's or stand-alone application servers in an "Applications & Services" network cloud.
  - ⇒ As illustrated in the figure, the IMS will mandatory interconnect to various other networks but support for other access networks than GERAN/UTRAN is not part of the Rel. 5 recommendations.

- **User Terminals**
  - ⇒ The range of user terminals is wide and includes legacy analog telephones as well as multi-mode PDA's, supporting various different IP-CAN's. This depends on customer preferences and on the operator type (fixed line telephone companies need to slowly migrate their analog users to the new world and therefore they need to support these analog user terminals at least medium term).
  - ⇒ In that respect, the type of user terminal also determines or limits the service types that a particular user is able to access. Obviously, an analog PSTN-terminal is unable to support video calls or instant messaging.

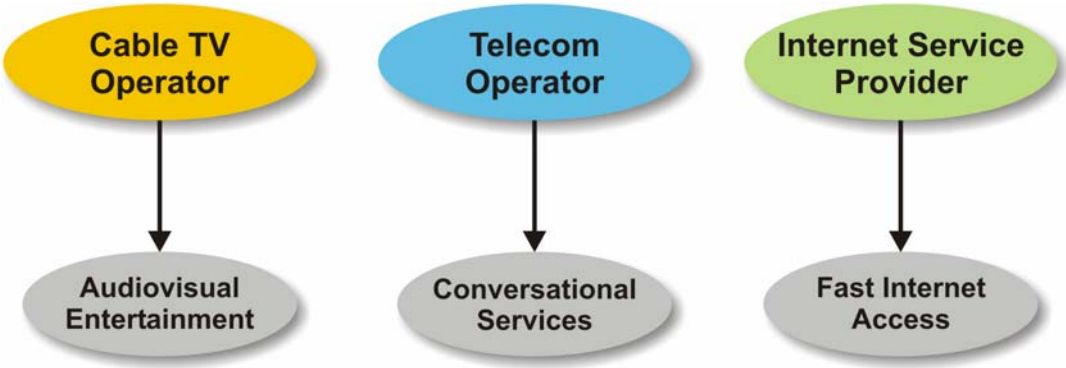
Room for your Notes

- **Abbreviations of this Section:**

CAN	Connectivity Access Network	**NAPT**	Network Address Port Translation (RFC 3022)
GERAN	GSM EDGE Radio Access Network	**NAT**	Network Address Translation (RFC 1631)
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**PSTN**	Public Switched Telephone Network
IP	Internet Protocol (RFC 791)	**UTRAN**	UMTS (Universal Mobile Telecommunication System)Terrestrial Radio Access Network
IP-CAN	Internet Protocol - Connectivity Access Network (e.g. DSL, TV-Cable, WIMAX, UMTS)	**PDA**	Personal Digital Assistant

1.3.2 Triple Play and Quadruple Play

1.3.2.1 Initial Situation



The objective of this section is to illustrate different service types as a main business domain of different operator types.

The key point of this section is that the ISP's started out very early, with the advent of DSL, to encourage or at least allow their customers to use VoIP to experience cheaper telephone calls. In that respect, ISP's were the first ones to cannibalize part of the business of another type of operator.

- **Cable-TV Operators**
  Historically, cable TV operators have been focusing on the provision of audio visual entertainment services through their HFC-infrastructure (Hybrid Fiber- / Coaxial-cable). The business model was originally based on broadcast services like television and standard radio exclusively. This explains two inherent assets of cable TV operators: They frequently lack capacity in the upstream direction (from user to network) and they prefer flat rate charging models.

- **Telecom Operators**
  The genuine business of telecom operators is the provision of conversational services to clients. The dominant conversational service always has been plain voice telephony. The telecom operators are historically the oldest operator type shown here: They are around since about 100 years. It was only one or two decades ago that market liberalization encouraged the spin off of mobile telecom operators and the foundation of new and independent fixed and mobile telecom operators with more or less market success.

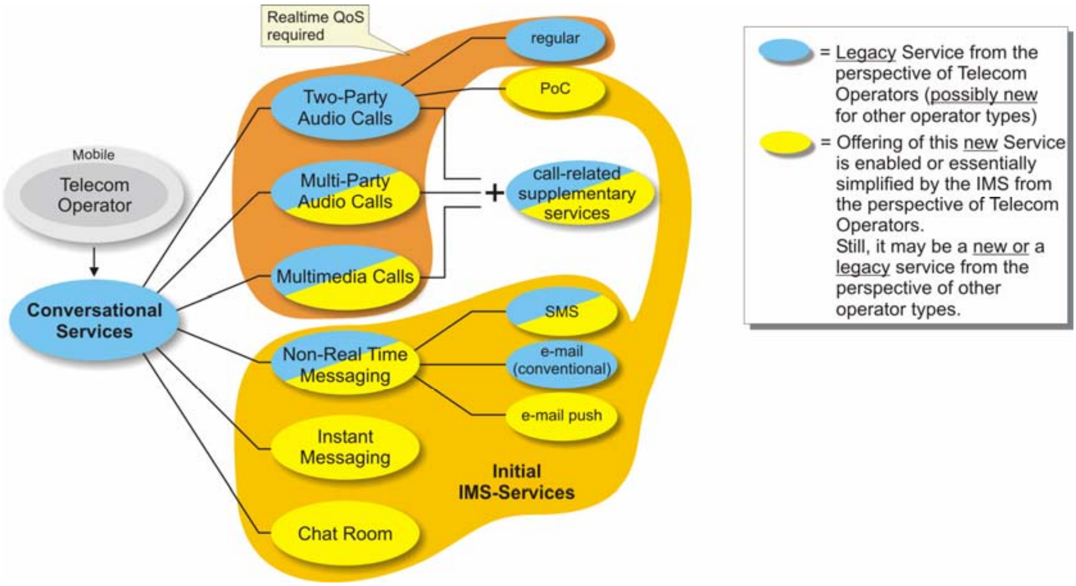
- **Internet Service Provider**
  Historically, the ISP is the “new kid on the block” compared to the two other types of operators. Originally, the exclusive business of the ISP was interfacing residential and business customers to the internet or even making these customers part of the internet. In that respect, the majority of customers (at least almost all residential customers) were interconnected to the internet through dial-up modem connections that required the presence of a telephone line which however is owned by the telecom operator. Basically, this is still true although many if not most residential users today use DSL over their telephone lines to interconnect to their ISP. Still, the disadvantage remains: ISP’s usually suffer from the lack of an independent access network between themselves and their customers. This is a serious drawback when it comes to the chances of becoming a successful triple-play-services provider.

Room for your Notes

- **Abbreviations of this Section:**

DSL	Digital Subscriber Line	**TV**	Television
ISP	Internet Service Provider	**HFC**	Hybrid, Fiber-, Coaxial- Cable
VoIP	Voice over Internet Protocol		

1.4.1 Conversational Services



The objective of this section is to depict the various different conversational services that may be offered through the IMS.

Key points of this section are:
1. The distinction between legacy conversational services (blue) and new conversational services (yellow) from the perspective of telecom operators.
2. The distinction between conversational services that do require real-time QoS and those do not.

Conversational services are the domain of telecom operators. Please note that the telecom operator may be a regular one or a mobile network operator.

- **Two-Party Audio Calls**
  As illustrated, this service type also contains PoC, because PoC is no longer a proprietary service when offered through the IMS.
  Note that (mobile) telecom offer close to perfect two-party audio call services without the IMS. It needs to be emphasized that for them the focus of the IMS therefore should not be on the provision of this kind of services, esp. initially. At a later stage, audio calls etc. may be migrated to the IMS to use only one service delivery platform.

- **Multiparty Audio Calls**
  The bullet is colored partly yellow because the conduction of multiparty calls becomes much more common, inherent and easy to use with the IMS.

- **Call-Related Supplementary Services**
  Any IMS-solution has to continue offering the legacy call-related supplementary services like caller representation, call on hold or call forwarding. However, the IMS will add new call-related supplementary services like “black lists” for callers or simultaneous forking.

- **Multimedia Calls**
  Through the IMS, the setup and selection of video calls will be simplified (although video calls have been around for quite some time). In addition, the IMS offers the combination of more media types than just audio + video. For instance, a service offering may combine audio + whiteboard + instant messaging to provide for advanced audio conferencing. Yet another example for multimedia calls is a “see what I see” service.

- **Non-Real Time Messaging**
  This bullet is colored partly yellow because some form of non-real time messaging has been there also prior to the IMS. However, SMS is (usually) a new service type for wire line telecom operators and definitely the famous e-mail push service is new.

- **Instant Messaging**
  Instant messaging has become an important means to communicate but it is a new type of service for telecom operators (both fixed and mobile).

- **Chat Rooms**
  The same applies what was said for instant messaging.

> Please note the remarks on the right hand side of the graphics page. The color for new and legacy services only applies for telecom operators but not necessarily for other operator types, offering triple play services. One example is instant messaging which is a new service for telecom operators but which definitely is a legacy service from the perspective of ISP’s.

- **Abbreviations of this Section:**

IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**QoS**	Quality of Service
ISP	Internet Service Provider	**SMS**	Short Message Service (3GTS 24.011, 3GTS 23.040)
PoC	Push to talk over Cellular (3GTR 29.979 and various OMA-specifications)		

1.5.1 See What I See

The objective of this video clip is to provide understanding about one possible usage of the “See what I see” application which is based on video telephony.

Video Sharing is a new multimedia service that enriches 3G mobile users' communication by allowing them to share a live camera view or a video clip while speaking on their mobile.

The video clip can be downloaded through the internet. Please check with your vendor.

- **“See What I See” Application**
  - ⇒ Mobile video sharing is currently in the “hot spot” because it can be deployed immediately as a combinational service (combining circuit switched and packet switched services) with existing circuit-switched calls, allowing you to launch and deploy a new multimedia service without cannibalizing existing voice revenues.
  - ⇒ Video streams are captured by the phone's camera or simply taken from video files stored in the phone's memory. It provides the users with an instant way of adding a visual element into a phone conversation.
  - ⇒ Video Sharing is based on standardized 3GPP and IETF technologies.

⇒ Mobile Video Sharing is a groundbreaking service that delivers fast, fun, and simple real-time communication.

Room for your Notes

- **Abbreviations of this Section:**

3GPP	Third Generation Partnership Project (Collaboration between different standardization organizations (e.g. ARIB, ETSI) to define advanced mobile communications standards, responsible for UMTS)	**IMS**	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)
IETF	Internet Engineering Task Force (www.ietf.org)		

1.5.2 Intelligent Address Book (Presence Service)

The objective of this video clip is to provide understanding about the usage of the “Intelligent Address Book” application which is enabled though presence service.

The key points of this video clip are:
1. In “Intelligent Address Book” application, the presence service lets end-users decide which important information related to presence they want to provide to a list of authorized contacts.
2. The illustrated graphics only represents the first snapshot of the presented movie.

The video clip can be downloaded through the internet. Please check with your vendor.

- **“Intelligent Address Book” Application**
  ⇒ From an address book containing live presence information you can choose the best mean of communication(e.g. voice, video and instant messaging) to reach your community partners, whether they are on their move or at their PC’s regardless of their access solution they rely on; i.e. GERAN/UTRAN, WIMAX, DSL, WiFi etc

⇒ The presence service is able to indicate whether other users are online or not and if they are online, whether they are idle or busy, e.g. attending a meeting or engaged in phone call.

⇒ Additionally, the presence service allows users to give details of their communication means and capabilities, e.g. whether they have audio, video instant messaging etc.

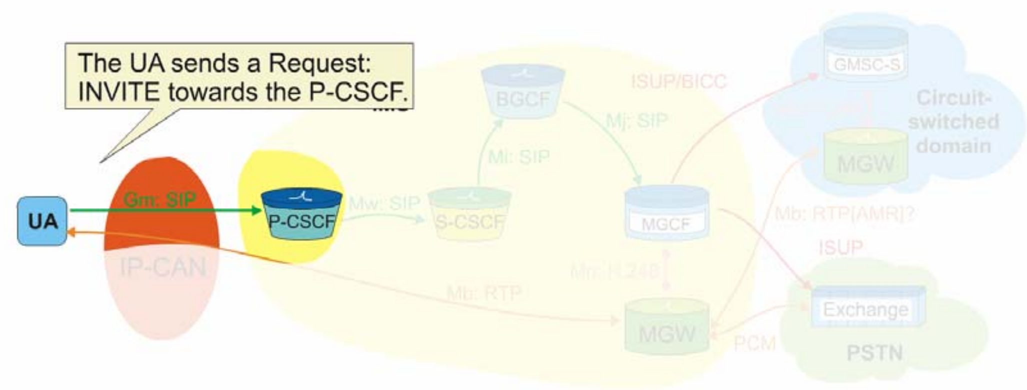
Room for your Notes

- **Abbreviations of this Section:**

DSL	Digital Subscriber Line	**UTRAN**	UMTS (Universal Mobile Telecommunication System)Terrestrial Radio Access Network
GERAN	GSM EDGE Radio Access Network	**WIMAX**	Worldwide Interoperability for Microwave Access (IEEE 802.16)
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**PC**	Personal Computer

1

1.6.1 IMS-Originating Voice Call – Technical Realization (ctd)



The objective of this section is to illustrate the next snapshot of the illustrated video clip.

Key point of this section is to illustrate how the IMS-UA initiates the voice call establishment by sending a SIP: INVITE-message towards the P-CSCF. This message identifies the called party.

The video clip can be downloaded through the internet. Please check with your vendor.

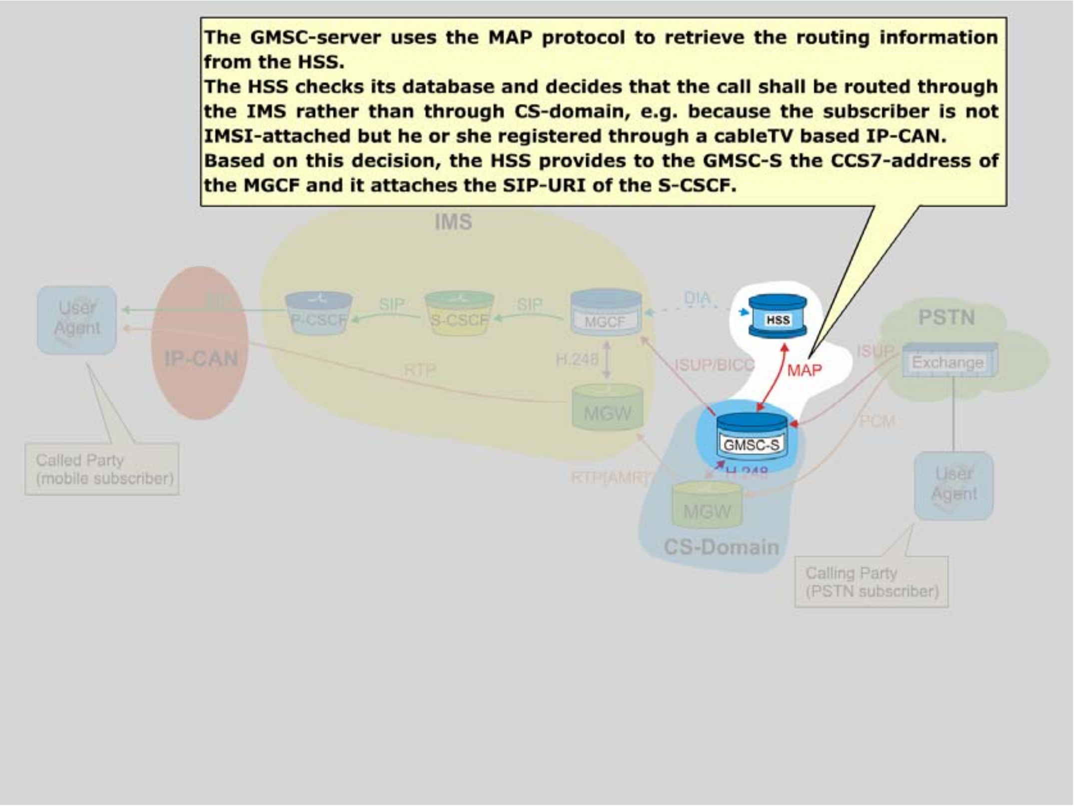
Room for your Notes

- **Abbreviations of this Section:**

AMR	Adaptive Multirate Encoding (3GTS 26.090)	**MGCF**	Media Gateway Control Function
BGCF	Breakout Gateway Control Function	**MGW**	Media Gateway
BICC	Bearer Independent Call Control (ITU-T Q.1902.1 - Q.1902.6)	**P-CSCF**	Proxy Call Session Control Function (SIP)
GMSC-S	Gateway MSC Server	**PCM**	Pulse Code Modulation
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**PSTN**	Public Switched Telephone Network
IP	Internet Protocol (RFC 791)	**RTP**	Real-time Transport Protocol (RFC 3550, RFC 3551)
IP-CAN	Internet Protocol - Connectivity Access Network (e.g. DSL, TV-Cable, WIMAX, UMTS)	**S-CSCF**	Serving Call Session Control Function (SIP)
ISUP	ISDN User Part (ITU-T Q.761 - Q.765)	**SIP**	Session Initiation Protocol (RFC 3261)
		UA	User Agent (SIP-Term / RFC 3261)

1

1.6.2 IMS-Terminating Voice Call – Technical Realization (ctd)



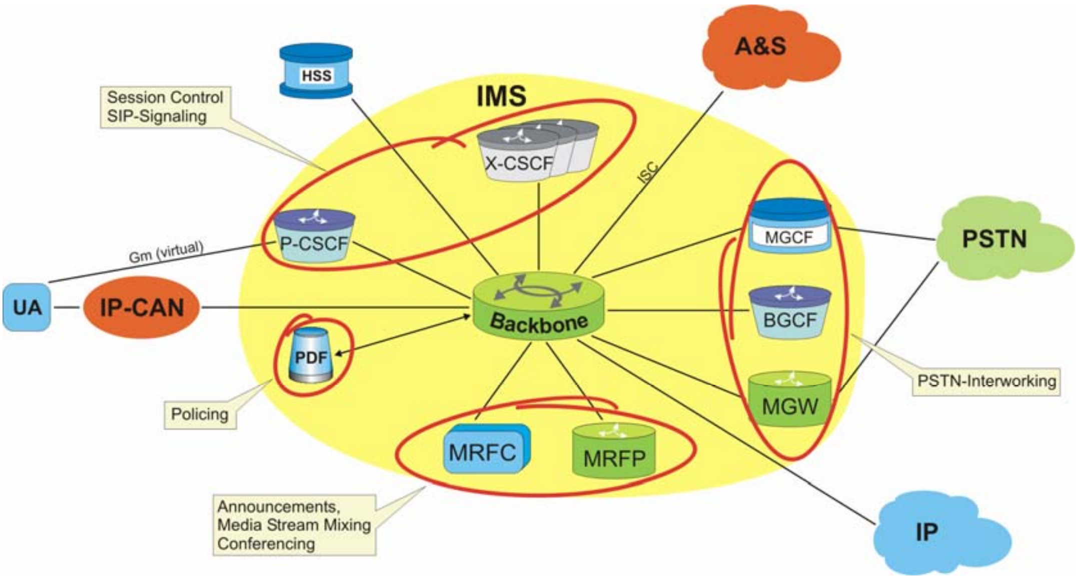
The objective of this section is to illustrate the next snapshot of the illustrated video clip.

Key point of this section is to illustrate that it is always the HSS that decides how the call routing shall continue. In this case, the decision is to route through the IMS.

The video clip can be downloaded through the internet. Please check with your vendor.

2

2.1 The IMS-Network Architecture



The objective of this section is to provide an overview of the logical network architecture of the IMS and its interconnections towards external networks.

The key points of this section are:
1. The grouping of the logical network elements of the IMS into three groups: X-CSCF, MRF and soft switches. The comprehension of these and the other functional groups is essential to understand the IMS, altogether.
2. The presentation of the IMS as mediator between IP-CAN and user on one hand and applications and services on the other hand.
3. This figure is to illustrate a simplified IMS (IP Multimedia Subsystem) in its environment. Focus is on the 3GPP-version of the IMS [3GTS 23.002, 3GTS 23.228] as we also included the HSS which is 3GPP-specific. The term "simplified" in the previous sentence means that not all internal relations and network elements within the IMS have been included.

- **With respect to the IMS-architecture please note the following:**
  - ⇒ P-CSCF and the other X-CSCF's represent SIP-servers with different functions. In that respect, the P-CSCF is the interface of the IMS towards user equipments (e.g. mobile stations).
  - ⇒ The PDF is used for policing (QoS).
  - ⇒ The MRFC and MRFP are used for announcements towards subscribers (e.g."This user is currently not reachable.")
  - ⇒ BGCF, MGCF and MGW are required for PSTN-interworking.
  - ⇒ There is no requirement that each logical node (X-CSCF, PDF, …) is represented by a separate physical node.

⇒ The Gm-interface between the UE and the P-CSCF is obviously a virtual interface that is physically realized through the packet-switched core network domain and the respective access network.

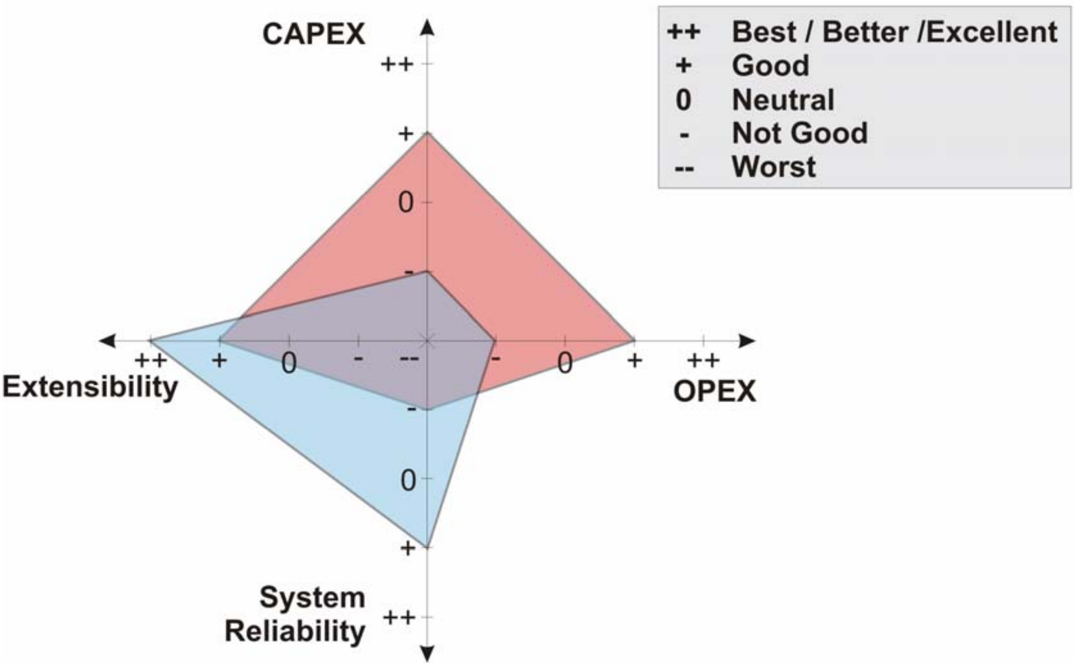
- **Abbreviations of this Section:**

3GPP	Third Generation Partnership Project (Collaboration between different standardization organizations (e.g. ARIB, ETSI) to define advanced mobile communications standards, responsible for UMTS)	**MRFC**	Multimedia Resource Function Controller
3GTS	3rd Generation Technical Specification	**MRFP**	Multimedia Resource Function Processor
BGCF	Breakout Gateway Control Function	**P-CSCF**	Proxy Call Session Control Function (SIP)
CAN	Connectivity Access Network	**PDF**	Policy Decision Function (Part of the IP Multimedia Subsystem)
CSCF	Call Session Control Function (SIP)	**PSTN**	Public Switched Telephone Network
HSS	Home Subscriber Server (3GTS 23.002). HSS replaces the HLR with 3GPP Rel. 5	**QoS**	Quality of Service
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**SIP**	Session Initiation Protocol (RFC 3261)
IP	Internet Protocol (RFC 791)	**UA**	User Agent (SIP-Term / RFC 3261)
IP-CAN	Internet Protocol - Connectivity Access Network (e.g. DSL, TV-Cable, WIMAX, UMTS)	**UE**	User Equipment
ISC	IP multimedia subsystem Service Control-Interface	**X-CSCF**	Call Session Control Function (any, there is I-CSCF, P-CSCF and X-CSCF)
MGCF	Media Gateway Control Function	**MRF**	Multimedia Resource Function
MGW	Media Gateway		

2

2.1.2 Comparison between Centralized and Split Architecture Approaches

Approaches \ Comparison Parameters	CAPEX	OPEX	System Reliability	Extensibility
Centralized Architecture	+	+	−	+
Split Architecture	−	−	+	++



The objective of this section is to compare two different IMS architecture implementation approaches with respect to different system evaluation parameters.

The key points of this section are:
1. With respect to CAPEX and OPEX, the centralized architecture approach to implement IMS in NGN's is better than split architecture approach.
2. With respect to System Reliability and Extensibility, the split architecture approach to implement IMS in NGN's is better than centralized architecture approach.

- **CAPEX**
  Capital expenditures are expenditures used by a company to acquire or upgrade physical assets such as equipment, property, industrial buildings. It refers to the cost of developing or providing non-consumable parts for the product, equipment or system.

- **OPEX**
  Operational expenditures are the on-going costs for running the product, equipment, business, or system. It is the counterpart of CAPEX. For example, the purchase of equipment or network components of telecom systems is the CAPEX, and the cost of different means which are used to run that equipment such as the cost of workers, electricity, facility expenses like rent and utilities are OPEX.

- **System Reliability**
  Reliability is the ability of a system to perform and maintain its functions in routine circumstances, as well as hostile or unexpected circumstances. It is the probability that the system will perform required functions for a specified period of time under stated conditions. It is also defined as the resistance to failure of a system and the capacity of a system to perform as designed.

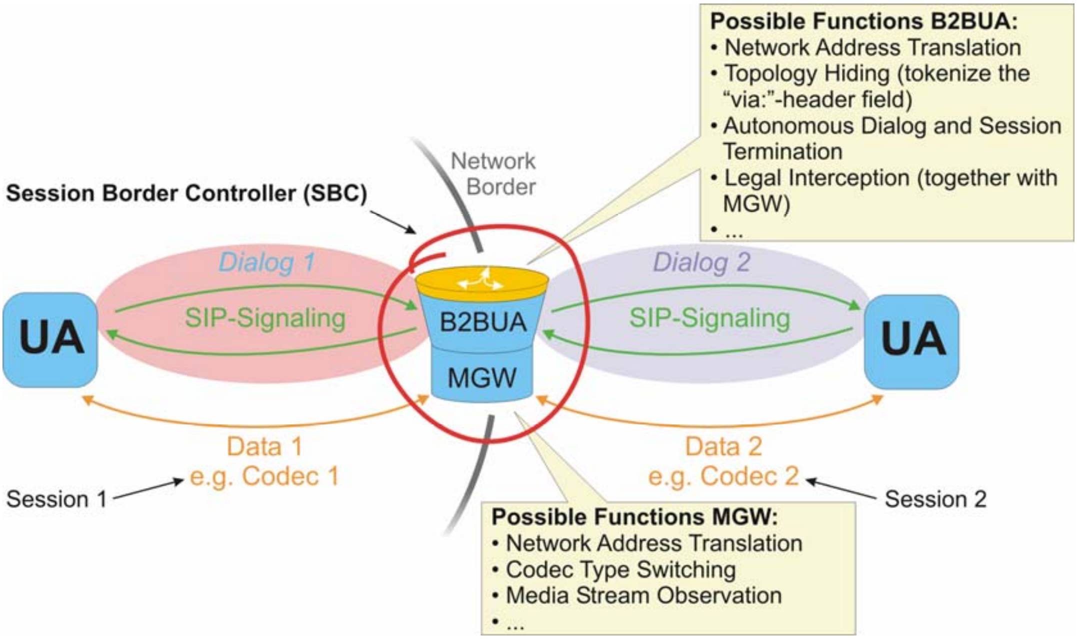
- **Extensibility**
  Extensibility is a systemic measure of the ability to extend a system and the level of effort required to implement the extension. It is a system design principle where the implementation takes into consideration future growth. In Systems Architecture, extensibility means that the system has been so architected that the design includes all of the hooks and mechanisms for expanding the system with new capabilities without having to make major changes to the system infrastructure.

- **Abbreviations of this Section:**

CAPEX	Capital Expenditure	**NGN**	Next Generation Networks
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**OPEX**	Operational Expenditure

2

2.2.5 Detailed Consideration of SBC and B2BUA



The objective of this section is to illustrate the operation and some applications of an SBC.

The key points of this section are:
1. The SBC has two-folded nature. It consists of a B2BUA plus a media gateway for data mediation.
2. Regularly, the B2BUA establishes two apparently independent dialogs between two user agents (dialog 1 and dialog 2).

The following list of functions of the B2BUA is by definition not exhaustive:

- **Network Address Translation**
  NAT allows using private IP-addresses network internally. Hence, the B2BUA will replace all internal IP-addresses by its own public IP-address. We will later talk in detail about SIP and NAT. Note that with respect to the NAT-function, the IETF uses the term ALG (Application Layer Gateway).

- **Topology Hiding**
  Operators may want to hide the internal network structure from externals. This relates particularly to the inherent exposure of SIP-proxy host names within the "Via:"-header field. The B2BUA will simply tokenize all previous entries in the SIP-header field before a SIP-message is forwarded to an external destination.

- **Dialog and Session Termination**
  Operators have a need to be able to intercept and possible even interrupt ongoing dialogs and sessions (e.g. if a prepaid service expires). We will later illustrate the possible application of this in more detail.

- **Autonomous Dialog Management and SIP-Parameter Alteration**
  This means that the B2BUA generates SIP-Requests and even entire dialogs autonomously. Also not possible for a SIP-proxy server is a reduction of the “expires”-parameter in case of registrations of user agents which enforce a more frequent re-registration. This is standard operational behavior for B2BUA’s.

And, in the media gateway:

- **Network Address Translation**
  The SBC enforces the media streams to be routed through itself. Additionally, NAT allows for the internal use of private IP-addresses.

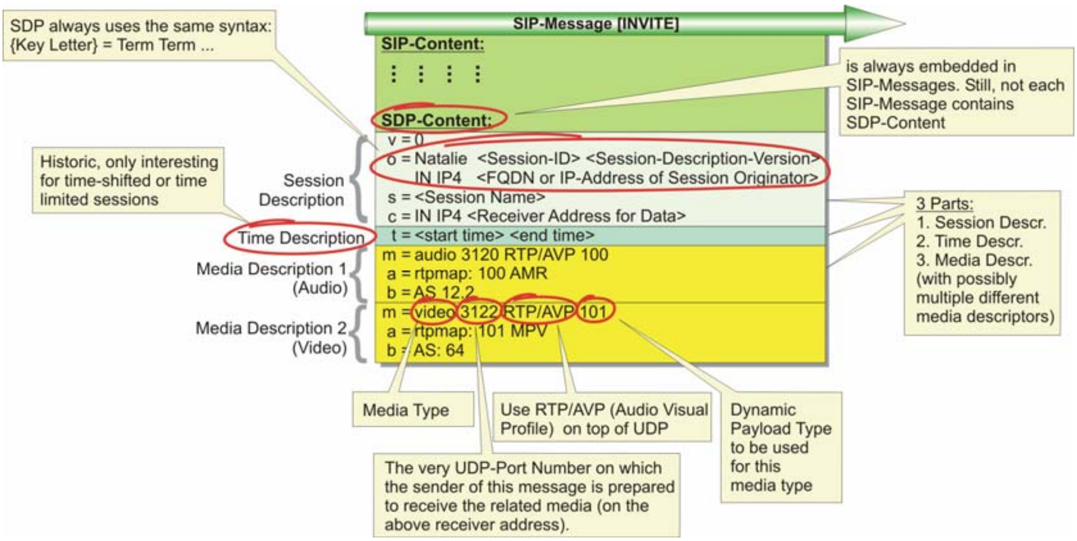
- **Code Type Switching**
  Operators are able to restrict the use of media codec’s to only a few types and they can convert to a different codec type before the data is relayed.

- **Media Stream Observation**
  Operators may want to observe media streams (e.g. for legal interception or for ungraceful session release).

- **Abbreviations of this Section:**

ALG	Application Layer Gateway	**NAT**	Network Address Translation (RFC 1631)
B2BUA	Back-to-Back User Agent (SIP term / RFC 3261, RFC 3725)	**SBC**	Session Border Controller (SIP term, usually a B2BUA with NAT-function and media gateway)
IETF	Internet Engineering Task Force (www.ietf.org)	**SIP**	Session Initiation Protocol (RFC 3261)
IP	Internet Protocol (RFC 791)		

2.3.4 Consideration of SDP



The objective of this section is to illustrate the principles of the Session Description Protocol.

The key points of this section are:
1. SDP manifests itself only within the body of SIP-messages. That is, SDP is nothing else but a set of information elements within the body of a SIP-message.
2. SDP-information consists of three parts: (1) Session Description elements; (2) Time Description Elements; (3) Media Description Elements.
3. The discussion of multiple parallel media streams is easy through SDP: Just add another m-line within the media description for each new media type.

The sequence of SDP-parameters within a SIP-message is always the same:

- **Session Description Items**
  This part contains several lines of ASCII-encoded information which contains overall, common session related information. This relates to global information about a session like the origin and the IP-address and IP-address type of the sender of this SDP. The username is “Natalie” and as presented, there is a session-ID and a session-description-version number allocated by this user. Besides, the session description contains the FQDN or IP-address of the user.
  The c-line specifically indicates on which IP-address Natalie is prepared to receive data.

- **Part 2: Time Description Items**
  This part contains start and end time of the session to be setup. However, this information usually redundant and only there for compliance reasons with the original SDP-specification. The time descriptors are there for historic purposes and stem from the time when SDP was introduced to invite participants in possibly different time zones for possibly time shifted telephone conference sessions. The call shall start immediately <start time = 0> and shall last infinitely <stop time> = 0. Obviously, the call will be terminated by other means when desired by either user.

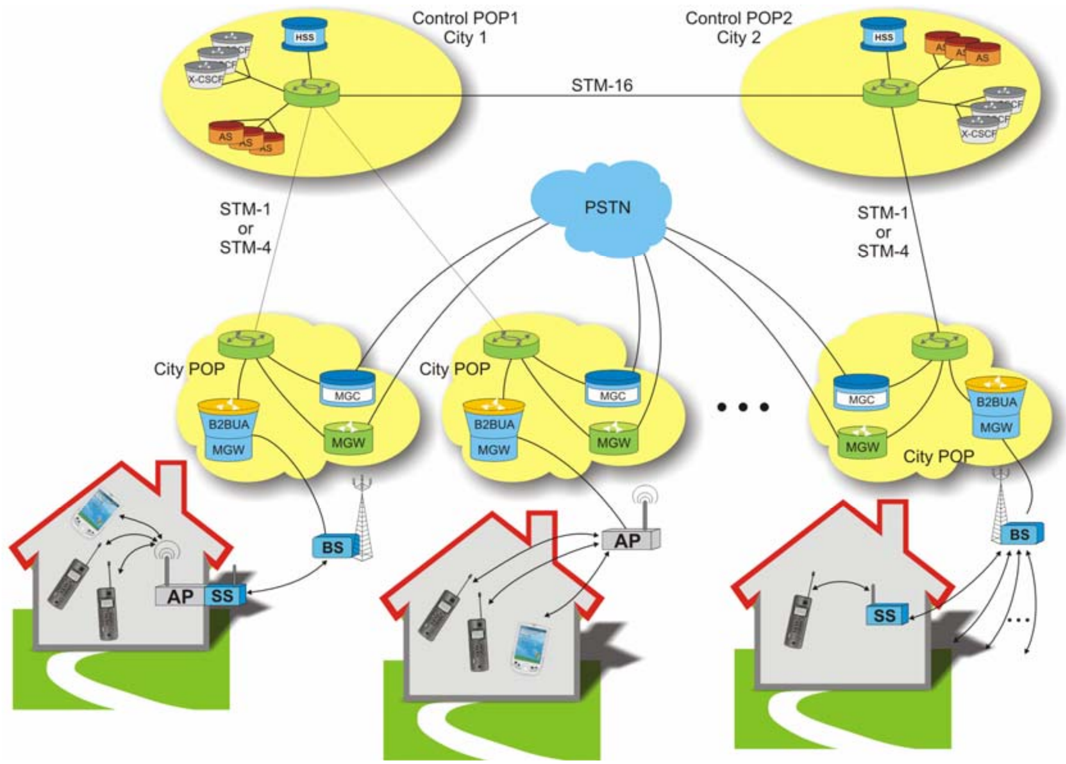
- **Part 3: Media Description Items**
  Two different media descriptors follow. Each is indicated by an m-item.
  ⇒ The first media description refers to the audio portion of the call. Natalie's host computer intends to use RTP/UDP/IP as transport protocol and the UDP-port number on the called party's computer where audio data will be sent is '3120'dec. Note that port number '3121'dec is reserved for the related RTCP-signaling.
  ⇒ They also identify as to which port number a media stream shall be sent.
  ⇒ Dynamic Payload type indication for RTP is selected. The RTP-Payload Type shall be '100'. Through the attribute line "a = rtpmap: 100 AMR", this dynamic payload type is assigned to the AMR-coder which has also been selected by Natalie on the user interface.
  ⇒ The b = AS: 12.2 tells the receiver that 12.2 kbit/s are required for the transfer of the AMR-data. The term 'AS' means that this bandwidth only relates to this media stream.
  ⇒ The second media description is related to the video portion of the call. As for the audio part, RTP/UDP/IP shall serve as transport protocol. The destination UDP-port number where the video data shall be sent is '3122'dec.
  ⇒ The dynamic RTP-Payload Type 101 is used. This Payload Type is assigned to MPV (MPEG-Video) in the line "a = rtpmap: 101 MPV.
  ⇒ Note how easy the description of multiple media streams for a single session is.

[RFC 2327, draft-ietf-mmusic-sdp-new-26.txt, http://www.iana.org/assignments /sdp-parameters]

- **Abbreviations of this Section:**

ASCII	American Standard Code for Information Interchange (ANSI X3.4-1986)	**SDP**	Session Description Protocol (RFC 2327, RFC 3266, RFC 3264)
IP	Internet Protocol (RFC 791)	**SIP**	Session Initiation Protocol (RFC 3261)
RFC	Request for Comments (Internet Standards)		

3.3 Physical Topology Design



The objective of this section is to illustrate the physical topology design model.

The key points of this section are:
1. There are two different parts in the network: Control POPs and City POPs.
2. Control POP's and City POP's together form the IMS.
3. Users are interconnected through WLAN and WIMAX-access links which terminate in SBC's.

- **Control POPs**
  Two main city nodes are connected to each other.

- **City POPs**
  The rest of eight cities are connected to different core nodes.

  ⇒ Project is done in phases to separate complexities and to minimize the CAPEX/OPEX costs in the initial stages of the network deployment.

The scope of phase approach is consists of the deployment of following elements:

⇒ WIMAX sites for Radio Access Network into the POP sites.
⇒ Transmission links from different WIMAX sites into the POP sites.

In order to provide service selection functionalities and billing capabilities, Post-paid & Prepaid, WAN Application Modules are installed in each of the redundant routers deployed in the Control POP sites.

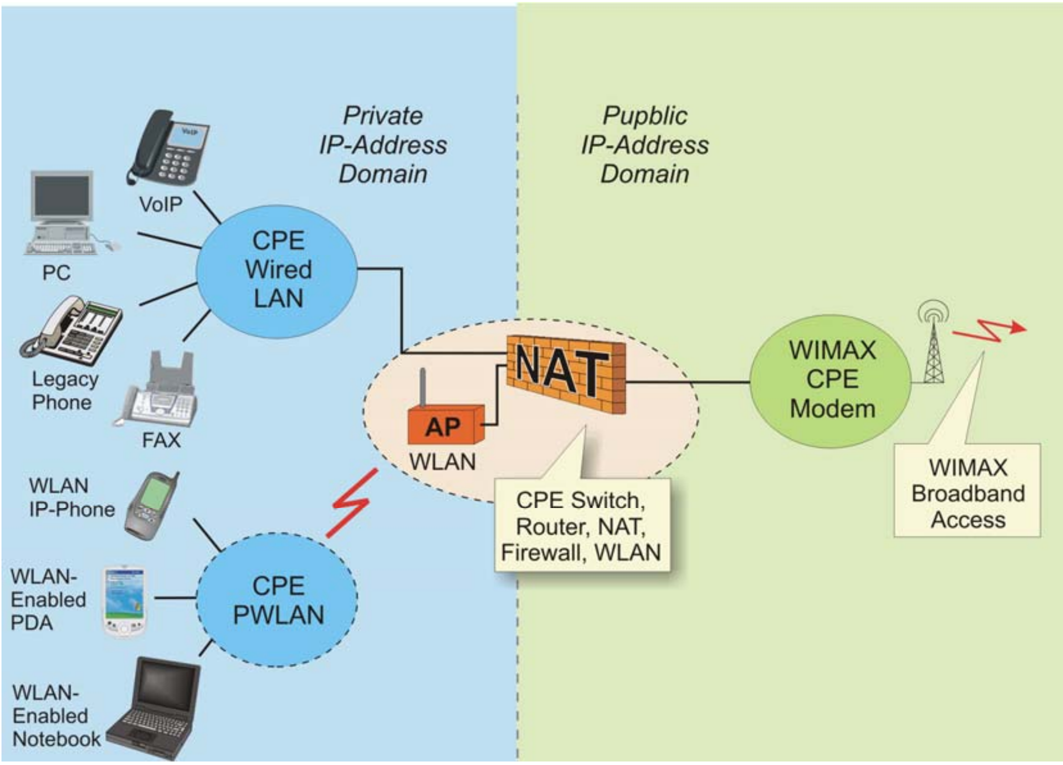
Room for your Notes

- **Abbreviations of this Section:**

AP	Access Point (IEEE 802.11, 802.16)	**OPEX**	Operational Expenditure
AS	Application Server	**POP**	Point Of Presence
B2BUA	Back-to-Back User Agent (SIP term / RFC 3261, RFC 3725)	**PSTN**	Public Switched Telephone Network
BS	Base Station (IEEE 802.16)	**SBC**	Session Border Controller (SIP term, usually a B2BUA with NAT-function and media gateway)
CAPEX	Capital Expenditure	**SS**	Subscriber Station (IEEE 802.16)
CSCF	Call Session Control Function (SIP)	**STM**	Synchronous Transfer Mode (SDH- / SONET-related)
DAP	Diversity Access Point	**ULAP**	Ultra Light Access Point
HSS	Home Subscriber Server (3GTS 23.002). HSS replaces the HLR with 3GPP Rel. 5	**WAN**	Wide Area Network
IMS	Internet Protocol Multimedia Core Network Subsystem (Rel. 5 onwards)	**WIMAX**	Worldwide Interoperability for Microwave Access (IEEE 802.16)
MGC	Media Gateway Controller	**WLAN**	Wireless Local Area Network
MGW	Media Gateway	**WLAN**	Wireless Local Area Network (IEEE 802.11)
		X-CSCF	Call Session Control Function (any, there is I-CSCF, P-CSCF and X-CSCF)

3

3.4 Customer Premises with WIMAX Access



The objective of this section is to provide an overview of the customer premise network architecture which is based on WIMAX access.

The key points of this section are:
1. Operator tends to create many wireless hot spots in different cities.
2. The architecture could be applied to both residential and business environments.
3. This is one possible option for wired and PWLAN end users to use WIMAX access.

⇒ A Public Wireless LAN or PWLAN is a region whereby users can access a network such as the Internet or company Intranet via a wireless enabled device providing they have sufficient authorization and authentication privileges to allow them access to the network. PWLANs are also known as WiFi zones, or hotspots.

⇒ The typical WiFi hot spot will feature then two to ten Access Points which are controlled by WLAN controllers.

3

- ⇒ This section describes the network design approach for a PWLAN access to be deployed by the operator in conjunction with IP MPLS core network with service provider voice capabilities.
- ⇒ The correct deployment configurations of standard equipment is needed for residential, small business, medium business, and large enterprise that allow, as much as possible, a "solution in a box" that is both easy to sell and easy to deploy.

Taken as separate functions, the following are the components which make up the customer premise network;

- ⇒ IP Broadband Network Access such as WIMAX
- ⇒ Customer Premise Home Gateway/Edge Router
- ⇒ End user terminals e.g. IP Phone (fixed), Soft Client and IP Phone (WLAN).
- ⇒

Room for your Notes

- **Abbreviations of this Section:**

IP	Internet Protocol (RFC 791)	**WIMAX**	Worldwide Interoperability for Microwave Access (IEEE 802.16)
NAT	Network Address Translation (RFC 1631)	**WLAN**	Wireless Local Area Network
RAN	Radio Access Network	**CPE**	Customer Premises Equipment