

# Multimedia Streaming Technology in 4G Mobile Communication Systems

Geeta Chandna, Mohit Bansal, Saloni, Saru Sehgal

*Abstract – Popularity and evolution of mobile devices like laptops, mobile phones and Personal Digital Assistants (P.D.A.), and the evolution of fast mobile networks in the last decade, have made it possible to increase the complexity of mobile applications and services provided to end-users. It is also a spectacular growth in multimedia communication especially via the World Wide Web. This paper explore some of the current technology of mobile devices, mobile networks and multimedia systems, and is based on the exploration outline some issues for design and development of mobile multimedia systems in 4G Mobile Communication System. Fourth-generation mobile communication systems will combine standardized streaming with a range of unique services to provide high-quality content (Multimedia) that meets the specific needs of the rapidly growing mobile market. By offering higher data-transmission rates up to 20 Mbps more than 3G for wide-area coverage and local-area coverage, 4G systems will be able to provide high quality streamed content to the rapidly growing mobile market.*

*Index Terms – 4G, Streaming, Code Division Multiple Access C.D.M.A. , Global System For Mobile (G.S.M.)*

## I. INTRODUCTION

Many portal sites offer streaming audio and video services for accessing news and entertainment content on the Internet from a PC. The term multimedia streaming means that there are more than one media type involved in the communication, e.g. text and graphics, voice, animations, video and audio. We define multimedia to denote the property of handling a variety of representation media in an integrated manner. This means that the various sources of media types are integrated into a single system framework. Currently, three incompatible proprietary solutions offered by Real Networks, Microsoft, and Apple dominate the Internet streaming software market. In the near future, third-generation mobile communication systems will extend the scope of today's Internet streaming solutions by introducing standardized streaming services, targeting the mobile user's specific needs.

By offering higher data-transmission rates up to 20 Mbps more than 3G for wide-area coverage and local-area coverage, 4G systems will be able to provide high quality streamed content to the rapidly growing mobile market. In addition to higher data rates, these systems also will offer value-added applications supported by an underlying network that combines streaming services with a range of unique mobile specific services such as Multimedia content, geographical positioning, user profiling, and mobile payment.

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Mobile cinema ticketing is one example of such a service. First, the mobile network or a terminal integrated positioning system such as G.P.S. would determine the user's geographical location. Then, the service would access a cinema database to generate a list of nearby movie theatres and a user profile database to determine what kind of movies the user likes best. Based on the geographical location information and user defined preferences, the service would offer the user a selection of available movies and show times. The user would then have the option of using the mobile device to view corresponding movie trailers through a streaming service. Upon choosing a film, the user could purchase a ticket through payment software on the mobile device. This and other mobile application scenarios present numerous challenges, such as how to provide spectrum efficient streaming services over varied radio-access networks to different types of end-user terminals. Our standard-based Interactive Media platform addresses these challenges by using an architecture that fits seamlessly into 4G mobile communication systems. An integral part of this architecture is a streaming proxy, which acts on both the service and transport levels. It is flexible enough to deal with different operator requirements and that it can provide high-quality streaming services in a mobile application environment.

## II. 4G MOBILE COMMUNICATION SYSTEMS

International Mobile Telecommunications – 2000 (IMT-2000) and the Universal Mobile Telecommunications System (UMTS) will be among the first 3G mobile communication systems to offer wireless wideband multimedia services using the Internet protocol. Two important technological changes will facilitate this advancement. The first change is a shift from last-generation radio-access technologies such as the global system for mobile (G.S.M.) communication, C.D.M.A. One (an IS-95 code division multiple access standard), and personal digital cellular (PDC) toward more sophisticated systems with higher data-transfer rates such as the enhanced data. Fourth-generation mobile communication systems will combine standardized streaming with a range of unique services to provide high-quality content that meets the specific needs of the rapidly growing mobile market. G.S.M. environment (E.D.G.E.), wideband C.D.M.A. (W.C.D.M.A.), and C.D.M.A.2000. As Fig. 1 illustrates, the second important technology shift is from a vertically integrated to a horizontally layered service environment. A horizontally layered 4G service network seamlessly integrates Internet protocol transport into a mobile service environment with a variety of access networks, opening up many new opportunities for IP-based mobile applications. For



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example, mobile terminals will be able to access existing Internet content through protocols and mark up languages such as WAP and WML that are optimized for wireless application scenarios. 4G mobile communications will have transmission rates up to 20 Mbps\_ higher than of 3G. The technology is expected to be available by the year 2010. 4G is being developed with the following objectives:

A. Speeds up to 50 times higher than of 3G. However, the actual available bandwidth of 4G is expected to be about 10 Mbps.

B. Three-dimensional virtual reality imagines personal video *avatars* and realistic holograms, and the ability to feel as if you are present at an event even if you are not. People, places, and products will be able to interact as the cyber and real worlds merge.

C. Increased interaction between corroborating technologies; the smart card in your phone will automatically pay for or will tell your car to warm up in the morning as your phone has noted you leaving the house. We can use new technology such as C.D.M.A. wireless access technology, advanced antenna

systems, next-generation mobile Internet, quality of service, power amplifier technology, and wireless access networks in 4G mobile communication system. 4G applications include high-performance streaming of multimedia content based on agent technology and scalable media coding methods. 4G will solve problems like limited bandwidth in 3G when people are moving and uncertainty about the availability of bandwidth for streaming to all users at all times. The 4G networks will also provide access to support services such as authentication, security, and billing mechanisms as well as mobile-specific services such as mobility management and location-based computing.

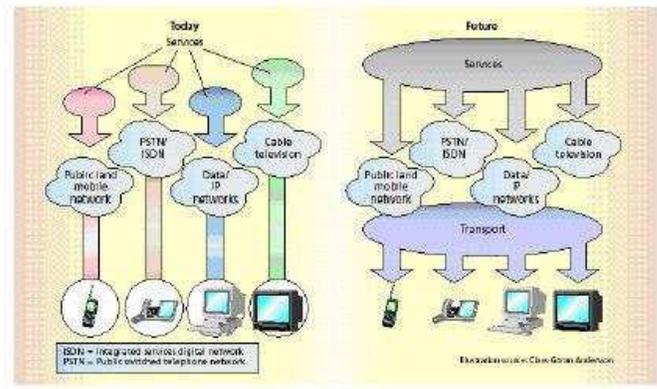
### III. MOBILE STREAMING CHALLENGES

The widespread implementation of mobile streaming services faces two major challenges: access network and terminal heterogeneity, and content protection.

#### A. Heterogeneity

In the future, we will have access to a variety of mobile terminals with a wide range of display sizes and capabilities. In addition, different radio-access networks will make multiple maximum-access link speeds available. Because of the physical characteristics of cellular radio networks, the quality and, thus, the data rate of an ongoing connection will also vary, contributing to the heterogeneity problem. One way to address heterogeneity is to use appropriately designed capability exchange mechanisms that enable the terminal and media server to negotiate mobile terminal and mobile network capabilities and user preferences. This approach lets the server send multimedia data adapted to the user's mobile terminal and the network. For example, a user accessing a specific service via a W.C.D.M.A. network could get the content delivered at a higher bit rate than someone using a general packet radio service or G.S.M. network. Similarly, when a person using a mobile multimedia terminal with a built-in low quality speaker plugs in a high-fidelity headphone, a dynamic capability exchange takes place, upgrading the transmission to

a high-quality audio stream for the remainder of the session. A related problem is how to efficiently deliver streamed multimedia content over various radio-access networks with different transmission conditions. This is achievable only if the media transport protocols incorporate the specific characteristics of wireless links, such as delays due to retransmissions of corrupted data packets. Here, proxies are a suitable approach for caching data packets and optimizing the data transport over the wireless links to a mobile terminal



**Fig. 1. The shift from a vertically integrated to a horizontally layered mobile service environment. 4G network seamlessly integrate Internet protocol transport with a variety of access networks.**

Video	Scene description	Presentation description
Audio	Presentation description	
Speech	Still images	
	Bitmap graphics	
	Vector graphics	
	Text	
Payload formats	Hybrid: tunable protocol	Real-time streaming protocol
Real-time transfer protocol		
User datagram protocol	Transport control protocol	User datagram protocol
Internet protocol		

**Fig. 2. The protocols integrate simultaneously playing video, audio, images, and formatted text into mobile multimedia applications**

#### B. Content protection

At the application level, controlling what users can do with content is an important challenge. The simplest form of content protection is simply disallowing the storage of received content. Content protection is part of the much broader digital rights management (D.R.M.) concept, which uses techniques such as encryption and conditional access based on usage rules to protect and manage access to multimedia data. Content providers are reluctant to deliver premium content over digital networks without D.R.M. mechanisms in place to prevent widespread illegal copying of valuable multimedia content such as music and movies.

#### IV. STREAMING STANDARDIZATION

Several organization and industry groups including the Internet Streaming Media Alliance (I.S.M.A.) and the Wireless Multimedia Forum (W.M.F.) have recognized the need for standardization of streaming services. Mobile streaming services in particular require a common standardized format because it is unlikely that mobile terminals will be able to support all proprietary Internet streaming formats in the near future. Using standardized components such as multimedia protocol stacks and codecs\_video and audio compression/decompression software in end-user equipment will help reduce terminal costs.

Furthermore, preparing and providing content in one standardized format is less time consuming and expensive than setting up content for several proprietary streaming solutions individually. We must to address mobile streaming standardization. Streaming services as an important building block of 4G multimedia applications. In addition to mobile streaming standardization, it is also require to addresses other applications such as videoconferencing and services for composing and receiving multimedia messages. Multimedia messaging services can include text, images, audio, short video clips, or video-stream URLs. We have to use the mobile packet-switched streaming service. This service integrates simultaneously playing video, audio, images, and formatted text into mobile multimedia applications.

The protocols and terminals for streaming applications are less complex than for conversational services, which require media input devices and encoders. There are some standard specifies both protocols and codecs. The protocols and their applications, illustrated in Fig. 3, are:

- 1) Real-time streaming protocol (R.T.S.P.) and session description protocol (S.D.P.) for session setup and control,
- 2) Synchronized Multimedia Integration Language (SMIL) for session layout description,
- 3) Hypertext transfer protocol (HTTP) and transmission control protocol (TCP) for transporting static media such as session layouts, images, and text, and
- 4) Real-time transfer protocol (RTP) for transporting real time media such as video, speech, and audio. The codecs and media types are
- 5) H.263 video,
- 6) MPEG-4 simple visual profile video (optional),
- 7) AMR (adaptive multirate) speech,
- 8) MPEG-4 AAC low complexity (AAC-LC) audio (recommended but optional),
- 9) JPEG and GIF images, and
- 10) XHTML-encoded, formatted text.

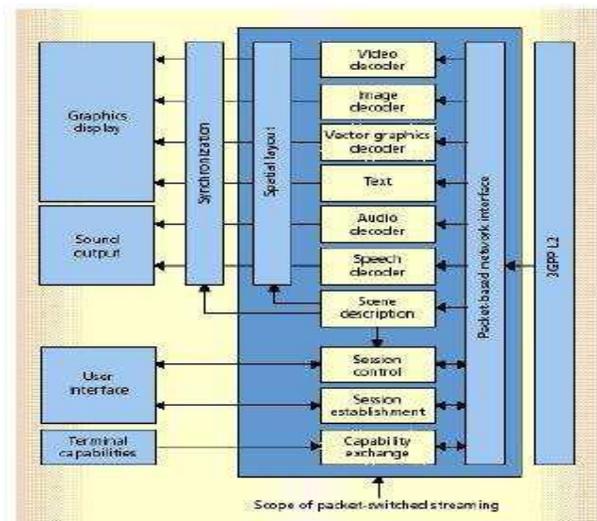


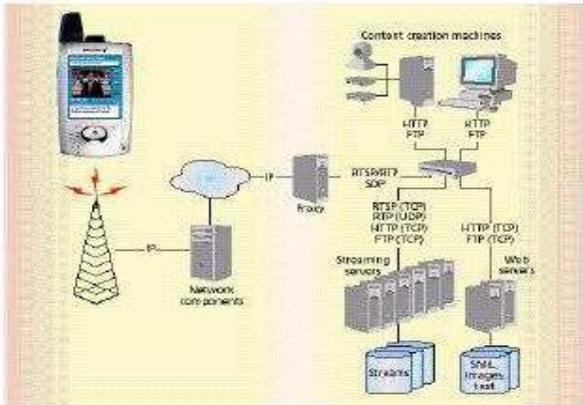
Fig. 3. Overview of streaming client

To enable interoperability between content servers, especially when inter working with MMS, the standard specifies using MPEG-4 as an optional file format for storing media on the server. The standardization process selected individual codecs on the basis of both compression efficiency and complexity. When combined using the SMIL presentation description language, the codecs enable rich multimedia presentations and applications, including video, audio, slideshows, and Multilanguage subtitling. Fig. 3 shows the logical components and data flow in a block diagram of a Streaming Standardization mobile-streaming terminal, including the individual codecs and presentation control. The network transmits the data and passes it to the application from Standard format link Layer. The application demultiplexes the data and distributes it to the corresponding video and audio decoders. The streaming standard offers the possibility of creating presentations in which several media elements such as video, audio, images, and formatted text play at the same time. SMIL, an XML-based presentation language developed by the World Wide Web Consortium, is the glue that combines these different elements to create an interactive multimedia presentation. SMIL is HTML with additional notions of time and temporal behaviour. Thus, it can describe a media screen and control the placement of media elements in space and time. The streaming client interprets the SMIL scene description and uses it to control the spatial layout and synchronization in the multimedia presentation. The standard specifically uses the SMIL 2.0 Basic Language Profile as well as the Event Timing, Meta Information, and Media Clipping modules. The additional modules add functionality such as changes in the presentation schedule based on user interaction (Event Timing), sending etainformation about the multimedia data (MetaInformation), and rendering only parts of a transmitted media stream (Media- Clipping). In addition, a streaming client can support the PrefetchControl module, which lets the content creator include hints about when to start a media stream.

## V. INTERACTIVE MEDIA PLAT-FORM

The Interactive Media system, illustrated in Fig. 4, is a software platform for mobile streaming applications. Designed as an end-to-end solution, the system consists of

- Dedicated content creation machines,
- A player application that runs on widely used operating systems such as Windows CE and EPOC,
- Content servers that hold the newly created multimedia content, and
- A proxy, which builds the interface between the player application and other parts of the platform.



**Fig. 4. Interactive Media platform.**

The client uses HTTP to request a SMIL presentation from a Web server. Within the SMIL presentation, the client finds links to the streaming content, which it acquires from the streaming servers. Static content, such as an image, is fetched from a Web server via HTTP. The chosen protocols are fully compliant with existing standards. HTTP provides access to static content through a TCP connection, while RTP packets transport streaming content via UDP connections. R.T.S.P. manages streaming sessions. As the Streaming standardization standard requires, the system uses S.D.P. via an R.T.S.P. connection to access stream descriptions. Introducing a proxy is necessary to fulfil the requirements of a mobile Internet application using off-the-shelf components designed for the fixed Internet. It also shields the core network from the back-end components and vice versa. Additionally, the back-end components can be located outside the operator domain, using the proxy with a firewall extension. This leads to a truly distributed architecture that puts the components into locations where they operate most effectively.

### A. Content Creation Machines

The content creation machines depicted in Fig. 4 host the applications needed for creating both live and offline content. They are used to prepare streaming content, for example, to edit videos and images and encode them in the appropriate formats for mobile streaming. Additionally, these machines create the SMIL files, which are a kind of storybook for the interactive presentation. They upload the content to the streaming servers for dynamic content and to the Web servers, which hold the static content and the SMIL files.

### B. Player Application

The player application renders multimedia content and lets users navigate through the SMIL presentations. Each multimedia element can be hyperlinked to other presentations. The player's SMIL implementation is fully standard-compliant as are the supported codecs, which decode multimedia data and render it on the output devices. Plug-in capabilities simplify extending the player with additional codecs. Applying skins changes the player applications appearance. A *skin* is a structure that adapts the look of an application's user interface. An application can have several skins. For example, a branding application implements the skin as images mapped on the side of the player's display and control elements. Selecting a different set of images for the skin brands the application for various customers. After launching the player application separately, the user can select a SMIL presentation or a single stream to navigate through a hierarchy of SMIL presentations. An alternative is to click a hyperlink in a standard Web browser that anchors a SMIL presentation. In either case, the player fetches the SMIL file from a Web server via the proxy. The player's SMIL engine interprets the contents of the SMIL file and fetches the streams (using the R.T.S.P. protocol) and the static content (using HTTP) according to the storyboard the SMIL file describes. The engine launches the content-specific codecs to render the information. Each of the elements in the SMIL file can have an underlying hyperlink. When the user clicks on a region of the screen that is associated with a hyperlink, the SMIL engine fetches the anchored file and interprets its content.

### C. Content Servers

Two kinds of back-end servers store the content the player renders: Off-the-shelf Web servers hold the SMIL pages, images, and other static content, and dedicated streaming servers store streaming content and related information. On reception of an HTTP GET request for an SMIL file from the proxy or the player, the Web server processes the request and fetches the appropriate content. Subsequent HTTP GET requests fetch the associated static content, or the user can click on a hyperlink in a SMIL file to fetch the new presentation. The player application uses the R.T.S.P. protocol to control the operation of the streaming server. After fetching the description of a streaming session, which it transports using the session description protocol, the player application sets up the streams of this session, for example, the video and audio track. When it receives an R.T.S.P. PLAY request, the server starts sending out RTP packets that transport the streaming content. Each stream can be in a different state for example, being set up, playing, paused. Therefore, the streaming servers must keep track of all active sessions. The server uses the real-time control protocol to provide the player, proxy, and streaming server with additional information about the session such as packet loss. Each stream that the server sends out has an RTCP connection.

### D. Proxy

The proxy is the system's interface to both the radio network and the back-end components.

This central component's major task is to adapt the streaming multimedia on the fly to the mobile network links continuously changing conditions. When a client requests an interactive multimedia presentation, the streaming proxy initially loads the SMIL file. The proxy's basic HTTP functionality optimizes the client connection according to the mobile IP network's characteristics. The client fetches the SMIL file and interprets it on the client, then the client requests both static and streaming content from the back-end servers. Acquiring static content such as images and text files is very straightforward, but the proxy's value becomes more apparent when it transmits streaming data to the client. During transmission of streaming data, the proxy dynamically adapts the delivered quality of service in accordance with available bandwidth. To achieve this dynamic adaptation, the proxy uses feedback information from the player application, radio network, and IP network. The user, content-provider, and operator use the proxy to configure preferences. A content provider can specify a minimum bandwidth to ensure acceptable video stream quality. If this bandwidth is not available, a slide show is presented instead. If the current bandwidth is insufficient for delivering a video, the proxy switches on the fly to a lower bandwidth as long as the QoS does not drop below a predefined value.

The operator also has the option of limiting bandwidth consumption to a certain user group, such as flat-rate subscribers. The system creates a reliable connection between the proxy and the client by retransmitting lost UDP packets for example, if the user passes through a tunnel and temporarily loses connection to the radio network. To provide streaming without losing information, the proxy automatically pauses the presentation when the connection is lost and then unpauses it after the terminal regains the network link. The proxy is also the interface to the operator's network components, including operation and maintenance, charging and billing, and subscription management. Mobile network operators can easily integrate the Interactive Media platform into existing structures and combine it with off-the-shelf products. The proxy itself is fully scalable at the machine level, using Telco standard load-balancing solutions when multiple machines use the same IP address. The proxy's major task is to adapt the streaming multimedia to the mobile network links continuously changing conditions.

## VI. CONCLUSION

Standardization of mobile services is being developed to overcome the challenges of streaming multimedia content in 4G mobile communication systems. The streaming standard specifies protocols and codecs that address streaming multimedia challenges such as the transmission characteristics of wireless links and the heterogeneity of radio-access networks and mobile terminals. The Interactive Media streaming platform, based on the standard, provides interfaces that application developers can use for charging and billing functions as well as network operation and maintenance. Ongoing developments concentrate on optimizing mobile content applications by supporting additional codes and offering a broader range of interfaces for proxy management and operation. Eventually, the platform will include extensions

for capability exchange to allow negotiation of terminal capabilities during session setup and digital rights management.

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