Recently, media convergence and interoperability over multiple networks is becoming a determining factor for the success of new digital broadcasting and telecommunication services. In this presentation, we propose an original architecture framework providing a mean for seamless transport of RTP based streaming over both Internet and ISDB (ARIB) networks. This goal is achieved by using the independence of the RTP protocol from the transport layer, and by remarking that MPEG-2 based systems such as ISDB provide a high-quality packet delivery scheme with the DSM-CC protocol. We propose to carry RTP/RTCP packets in DSM-CC messages, whereas an RTSP messages subset is being mapped onto ARIB DSM-CC event messages.

**Keywords**
Media convergence, multimedia streaming, ARIB, RTP, DSM-CC.

1. **INTRODUCTION**

Digital broadcasting (ARIB, DVB, ATSC) and telecommunication (Wireless, Internet) framework is entering a new era. We observe the multiplication of user profiles and demand for accessing the same multimedia applications on many different platforms (DTV, PDA, PC, car navigation systems and mobile phones), hooked seamlessly on several networks (broadband Internet, digital broadcasting or wireless 3G). Therefore, the need for a new interoperable mean for digital content delivery has been made clear. Several attempts have been proposed and/or implemented to bridge the gap between high QoS (Quality of Service) TV-centric digital broadcasting, static content oriented Internet and low bit-rate mobile phones. The Digital Video Broadcasting consortium (DVB) for example, has already standardised a way for carrying Internet Protocol packets over broadcast network (Ref. [1]) using DSM-CC addressable sections. However, this does not provide a fully satisfying solution for the goal of complete media convergence and interoperability.

In this presentation, we propose an original architecture framework providing a mean for seamless transport of Real Time Protocol (RTP, Ref. [2]) based streaming, over the Internet and digital broadcasting networks. RTP protocol is currently becoming the best candidate for streaming content delivery over the Internet as well as in future mobile phone networks. RTP has been initially introduced for Internet using UDP/IP as transport protocol. However, it is made by design to be transport agnostic and closer to the media application than to the details of the transport delivery. Therefore, we propose to use this characteristic to provide a common streaming protocol carried over Internet and broadcast network, such as BS or CS digital and later the digital terrestrial (ISDB-T).

We show in this paper, that RTP packets coming from a single source can be delivered in parallel over an Internet path and a digital broadcast path. To achieve this result, we encapsulate the RTP and RTCP packets in a modified form of DSM-CC sections, that are carried in MPEG-2 Systems Transport Stream (MPEG-2 TS). We then apply a mapping of a subset of RTSP messages (Ref. [3]) to DSMCC event messages, which ensures a minimal application level signalling interoperability.

Further conceptual details and implementation are discussed in a forthcoming publication (Ref. [4]).

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2. OVERVIEW OF FUNCTIONAL ASPECTS

2.1 RTP BASED STREAMING AND MEDIA CONVERGENCE

The Internet side

RTP based delivery of audio/video content is now becoming very popular in the Internet arena. It is used as the standard time-based packetization-method in many environments and sits just above transport layers, such as UDP/IP. RTP is associated with RTCP (Real Time Control Protocol) to add control and feedback capabilities to the delivery mechanism. An application-level protocol, RTSP which stands for Real Time Streaming Protocol (Ref. [3]), has also been developed to offer session negotiation and content description mechanism. RTP/RTCP and RTSP provide a standardised set of open protocols for real-time streaming of audio/video content. RTP/RTCP in principle, does not depend on the underlying transport protocol, and can be carried over other networks.

The digital broadcasting side

In principle, real-time multimedia delivery based on RTP/RTSP could be used for high-quality broadcasting, but the best-effort Internet does not offer enough transport reliability. Digital broadcasting environments (ARIB, DVB or ATSC) based on MPEG-2 system, target high-quality audio/video delivery with a guaranteed transport. MPEG-2 broadcasting networks assume a constant delay, a reduced jitter and packet loss, so RTP can be sent efficiently enjoying a high QoS for streaming applications.

Bridging the gap

Here, we propose a method to introduce RTP-based streaming-content in the digital broadcasting arena, in parallel to the conventional standards, providing a way to bridge the gap between non-fully interoperable environments. This strategy is motivated by the following aspects. First, as RTP is a ubiquitous and open standard, it could be used as a common packetized form for streaming content interchange. Audio/Video contents in RTP format can easily be sent over the Internet or stored on disk for later playing. Second, many new multimedia applications are entirely based on RTP for the streaming. The most appropriate example is the Java Media Framework (JMF, Ref. [5]) which has made RTP an entire part of its specification1. Applications based on Java and/or JMF are also entering the digital broadcasting and mobile phone areas. For example, DVB consortium has standardised a complete API called Media Home Platform (MHP, Ref. [6]) that includes JMF and Java as basic ingredients. Although the first release of MHP 1.0 does not include RTP streaming, we may expect that future releases will. In JMF 2.1, RTP streams are easily handled at the application level and the addition of decoder/encoder, packetizer/d depacketizer or renderer plug-ins sitting between the transport layer and the application is straightforward. This constitutes a great advantage when talking about interoperability and media convergence, hence the merit of delivering RTP packets in digital broadcasting networks. Finally, in many situations the path between the content producer and the consumer (the application that decodes/renders the RTP payload), is simplified when transport layer overhead is removed. For example, an RTP stream delivered over an MPEG-2 Transport stream do not necessitate the adjunction of the UDP/IP protocol overhead, as it would be the case if we chose to blindly proxy the RTP/UDP/IP packets using addressable sections as in reference [1]. The application can chose itself to consume the RTP stream or may decide to re-transmit by opening another UDP/IP channel. This concept is illustrated by the following figure.

![Figure 2-1 - RTP/DSM concept and media convergence](image)

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1 At least in release 2.0 onwards.
2.2 THE PROTOCOL LAYER ARCHITECTURE

The protocol layer architecture is designed in such a way as to make the distinction with the transport layer truly apparent. Considering a Producer (encoder) → Consumer (presentation) delivery path, the RTP packets are encapsulated in DSM-CC sections, without any Internet transport layer information such as port or IP address, and sent over the broadcast channel. The RTCP packets are encapsulated using the same method. Some S-R (Server-Client) RTSP messages can be usefully mapped in DSM-CC event messages and sent in the same stream. At the receiver side, the multimedia presentation is handled at the RTP level independently, and directly grabbed from the transport protocol, UDP or DSM-CC.

The protocol architecture is shown in Figure 2.2. The media stream coming out from the encoder is packetised using RTP format, and RTSP sender reports messages can be generated. This packetised stream is then either inserted in UDP packets or encapsulated in DSM-CC sections. UDP packets are usually sent over an IP protocol network, whereas DSM-CC sections are further packetised in an MPEG-2 transport stream and sent over a broadcast network or via ATM.

At the receiving side, the RTP/RTCP packets are extracted from the network (either MPEG2 data stream or UDP/IP) and presented to the application.

![Protocol Architecture Diagram](image)

Figure 2.2 - Protocol architecture for the transport of RTP packets over DSM-CC sections

The details of the RTP/RTCP packets encapsulation are given in a Ref. [4].

3. APPLICATION LEVEL SIGNALLING

3.1 ACCESS FROM MPEG2-TS SI

In UDP/IP based delivery scheme, a packet would be identified by the (network address, port number) pair. In broadcasting terms, this information has to be mapped to MPEG-2 System descriptor format. First, the RTP stream, once encapsulated in DSM-CC sections will be packetized in TS packets and a PID will be assigned to them. The way the receiving application can associate the PID to the RTP stream is achieved by extended the Program Map Section (PMT, Ref. [7]). The receiver would monitor the PMT and determine, first if a data broadcasting channel is currently available (stream_type = 0xD), then parse the descriptors and extract the component_tag/PID and RTP specific information carried in the newly defined RTP_descriptor. This process is summarised on Figure 3-1. As an example, the video on PID 0x100 and component_tag=94, the audio on PID 0x101 and component_tag=95, etc., would correspond to an original RTP stream like rtp://224.0.0.1:1994/video/1 and rtp://224.0.0.2:1994/audio/1.
3.2 ACCESS FROM PRESENTATION APPLICATION

The example of ARIB BML

In BS-Digital, multimedia presentations are encoded in BML format and the data sent over the MPEG-2 System stream using DSM-CC data carousel (Ref. [8]). We propose to define a new data source format to support delivery of RTP packets over DSM-CC. We define the data source format as:

```
rtpdsm://[...]/component_tag/
```

The [...] represents extra information found in the BS-Digital name space. As an example, taking an RTP stream broadcast on PID 0x100 and 0x101, with component_tag = 94 and 95 respectively the BML code would look like:

```
<obj id="Vstream" type="video" x-arib-mpeg4" data="rtpdsm://94"
    remain="remain" style="left:543px; top:30px; width:360px; height:202px"/>
<obj id="Astream" type="audio" x-arib-mpeg4" data="rtpdsm://95" remain="remain"
    streamstatus="play"/>
```

```
Figure 4.2 - BML code for accessing the RTPDSM data source format
```

Specific event signalling

In the BML context, the aforementioned RTSP to event mapping could be used in the following way:

```
<bevent>
  <beitem id="RTSPmessage" type="ANNOUNCE"
    subscribe="subscribe"
    onoccur="ProcessRTSPAnnounceMessage()"
  />
</bevent>
```

The ProcessRTSPAnnounceMessage() function could be a call to a downloaded agent (java class) that would handle the RTSP message properly in the context of the receiver and perform the required response to the server if a return channel is available.
4. Conclusion

In this presentation, we introduce an innovative solution for achieving media convergence between digital broadcasting and telecommunication networks. The basic concept relies on the use of the RTP protocol as a common streaming packet format, which is encapsulated in MPEG-2 DSM-CC sections when delivered over a broadcasting path or carried in UDP/IP packets when sent over the Internet. We showed that this concept solves many interoperability issues by making a clear distinction between the transport and application levels. We defined a new protocol named "rtpdsm" to access the streaming content in digital broadcasting receivers, as well as a convenient application signalling adapted to ARIB framework. The implementation is straightforward in current set-top boxes.

The "rtpdsm" would best fit in an environment such as MHP (Ref [6]) where the Java Media framework plays a central role. RTP is not a part of the initial MHP, which is based on JMF 1.0. However, extensions based on the rtpdsm protocol and JMF 2.0 would greatly enhance the interoperability of MHP platform with telecommunication networks. MPEG-4 video/audio streams can be also easily carried using the rtpdsm protocol and delivered on digital terrestrial broadcasting using the same format as on mobile network or the Internet.

5. REFERENCES