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ITU-T Y.1900-series – Deployment models of N-screen services

ITU-T Y-series Recommendations – Supplement 43

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Supplement 43 to ITU-T Y-series Recommendations

ITU-T Y.1900-series – Deployment models of N-screen services

Summary

N-screen services are about enabling a user to view same media content on multiple user devices. The translation and adjustment of content formats is necessary when the formats used in each user device are different. In order to address the challenge, Supplement 43 to ITU-T Y-series Recommendations describes three kinds of deployment model of N-screen services when user devices use the different protocols, metadata and stream formats. The service requirements are also specified when the deployment model is applied for the support of N-screen services.

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ITU-T Y.1900-series – Deployment models of N-screen services

1 Scope

This Supplement describes three kinds of deployment model of N-screen services when user devices use the different protocols, metadata and stream formats. The service requirements are also specified when the deployment model is applied for the support of N-screen services. This Supplement specifies:

- deployment models of N-screen services;
- service requirements of N-screen services.

2 References

[ITU-T Y.1911] Recommendation ITU-T Y.1911 (2010), *IPTV services and nomadism: Scenarios and functional architecture for unicast delivery*.

[ITU-T Y.2701] Recommendation ITU-T Y.2701 (2007), *Security requirements for NGN release 1*.

[ITU-T Y-Sup.24] ITU-T Y-series Recommendations – Supplement 24 (2013), *ITU-T Y.2000-series – Supplement on N-screen service scenarios for fixed mobile convergence*.

3 Definitions

3.1 Terms defined elsewhere

This Supplement uses the following terms defined elsewhere:

3.1.1 media [b-ITU-T H.324]: One or more of audio, video or data.

3.1.2 Internet Protocol Television (IPTV) [b-ITU-T Y.1901]: Multimedia services such as television/video/audio/text/graphics/ data delivered over IP-based networks managed to support the required level of QoS/QoE, security, interactivity and reliability.

3.1.3 N-screen service [ITU-T Y-Sup.24]: An information sharing and content providing service on all network enabled devices (TV, mobile phone, tablet, PC, home appliances and cars) that has a seamless and convenient manner.

3.1.4 video on demand (VoD) [b-ITU-T Y.1910]: A service in which the end user can, on demand, select and view video content and where the end user can control the temporal order in which the video content is viewed (e.g., the ability to start the viewing, pause, fast forward, rewind, etc.).

3.2 Terms defined in this Supplement

This Supplement defines the following terms:

3.2.1 media content: Set of media components (e.g., audio, video, timed text) that have a common timeline as well as relationships on how they may be presented (for example individually, jointly, or mutually exclusive) with an example being a program or a movie.

3.2.2 media presentation: Structured collection of data that establishes a bounded or unbounded presentation of media content composed of components of continuous media.

3.2.3 media presentation description: Formalized description for a media presentation.

3.2.4 segment: A resource that can be identified by an HTTP-URL and possibly a byte-range, included in the media presentation description (MPD), which when this entire resource thus identified is requested through HTTP/1.1 GET method (or partial GET with the indicated byte range) as defined in [b-IETF RFC 2616] the segment is the entity body of the request response.

3.2.5 streaming service: A service that the stream is constantly received by and presented to an end-user while being delivered by a provider. An end-user can begin playing the content (such as a movie) before the entire content has been transmitted.

4 Abbreviations and acronyms

This Supplement uses the following abbreviations and acronyms:

3GP	3GPP file format
3GPP	third Generation Partnership Project
DASH	Dynamic Adaptive Streaming over HTTP
HAS	HTTP Adaptive Streaming
HDS	HTTP Dynamic Streaming
HLS	HTTP Live Streaming
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
IPTV	Internet Protocol Television
MPD	Media Presentation Description
MPEG	Moving Picture Experts Group
OIPF	Open IPTV Forum
OTT	Over-The-Top content
PSS	Packet-switched Streaming Service
QoS	Quality of Service
RTCP	Real-time Transport Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
RTT	Round-Trip Time
SDP	Session Description Protocol
UAProf	User Agent Profile
UDP	User Datagram Protocol
VoD	Video on Demand
XML	extensible Markup Language

5 Conventions

In this Recommendation:

The keywords "is required to" indicate a requirement which must be strictly followed and from which no deviation is permitted, if conformance to this Recommendation is to be claimed.

The keywords "is recommended to" indicate a requirement which is recommended but which is not absolutely required. Thus, this requirement need not be present to claim conformance.

6 Overview

Traditional linear TV services, broadcasting TV programmes at the offered time whoever is watching or not, were initially dominant, but other streaming services, e.g., video on demand (VoD), Internet protocol television (IPTV) or Internet streaming services, are becoming more prevalent in a more flexible way, such as anytime, anywhere [ITU-T Y.1911].

The growth of Internet content (e.g., video) as well as the rapid proliferation of a variety of devices, tend to increase network traffic more and more. Due to viewer demand for personalization and flexibility, a variety of content types with different formats are accordingly also increasing. In addition, a variety of user devices (e.g., TVs, PCs, tablets, smart phones) and access networks (e.g., fixed broadband, WiFi, 4G, 5G) are driving the widespread consumption of video content. Therefore efficient delivery methods should be devised in order to enable customers to access all types of content on any device and at any time, so called N-screen services.

[ITU-T Y-Sup.24] also describes a kind of N-screen service scenario. Consumers are today increasingly using multiple devices at the same time. To meet demand, multi-device platforms enable users to enjoy the same content or services seamlessly, irrespective of which device or medium is being used. The N-screen service, which has recently been gaining attention, falls into the broader concept of a multi-device service. The N-screen service is about enabling the user to use multiple devices, which means that it should be made up of integrated platforms for multi-devices. The core element of the N-screen service is a platform that mediates the use of content or services on multiple devices.

The ability to deliver content to multiple screens is rapidly becoming a compulsory requirement for providers of such services as over-the-top content (OTT) and IPTV, as their aim is to meet customer needs for personalization and flexibility to earn revenue through such new business models.

In general, a video-streaming service is served in a client–server model and in a centralized architecture. It means that a user device in end user side and a media server in network side are tightly coupled as a pair and dependent on each other. Therefore a media server is really allowed to deliver all kinds of content to relevant user devices due to the difference in protocols and formats, especially for the support of N-screen services. The dedicated systems in each difference may lead to duplicated investments in deployment as well as operations. This Supplement aims to address these challenges.

In this regard, this Supplement describes deployment models of N-screen services. The service requirements are also specified for the support of N-screen services.

7 Deployment models

Many types of device with different capabilities (e.g., screen size, resolution) are available on the user side and furthermore different players (e.g., HTTP live streaming (HLS), HTTP dynamic streaming (HDS)) run on them. They are implemented on vendor specific and even standards (e.g., Moving Picture Experts Group (MPEG), Third Generation Partnership Project (3GPP), Open IPTV Forum (OIPF)) that are fragmented in terms of protocols, metadata and stream formats. Each user device can handle a limited set of metadata and stream formats, and it may not properly display media content on the screen if a certain format is not supported. In this regard, three different deployment models possibly occur for the support of N-screen services: 1) coupling each user device with a particular media server individually according to media delivery capabilities; 2) consolidating all media delivery capabilities on one server; or 3) converting an unsupported format to a supported format on each user device.

It is assumed that the content source is encoded in ITU-T H.264 codec and carried in a MPEG-TS container to each media server. The silo model means that each device is connected to the respective media server and served by the server, as illustrated in Figure 7-1. For example, a user device (e.g., TV) using dynamic adaptive streaming over HTTP (DASH) can communicate with the respective media server supporting DASH and then the media content with a proper format is delivered and successfully displayed. If the corresponding media server does not exist, the device is not served. In the silo model, media content should be transmitted in a format that both a user device and a media server can recognize. This model is a quite simple, but the media server is always duplicated.

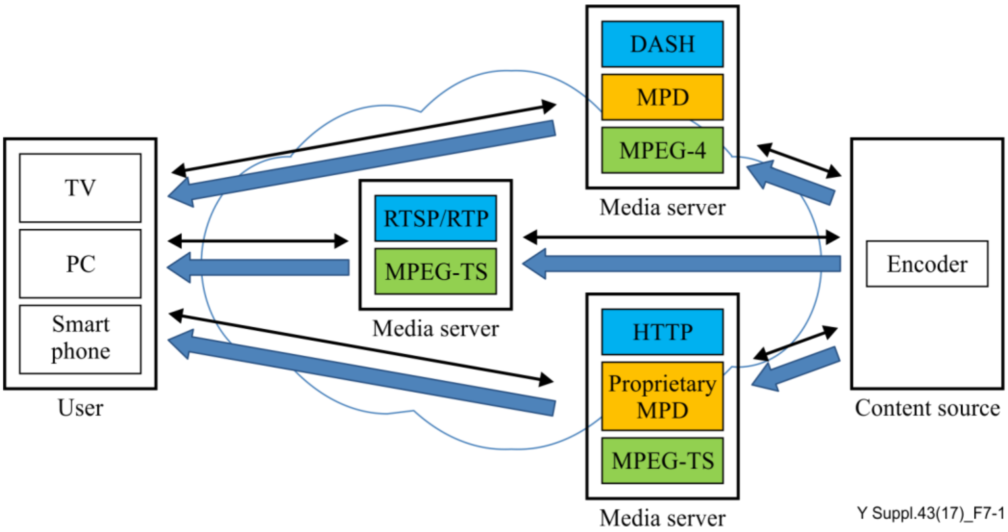

Figure 7-1 – Silo model

The consolidation model is represented by integrating all media delivery capabilities on one server, as shown in Figure 7-2. Therefore a media server is required to provide a variety of protocols, metadata and stream formats that every user device requires. In the consolidation model, a media server may be more complex and overwhelming than one in the silo model. The number of media servers, however, is fewer than that in the silo model.

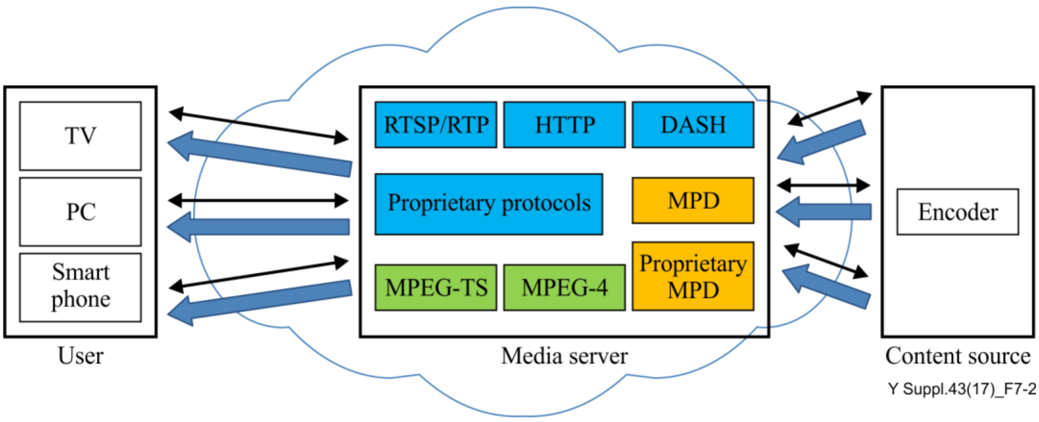

Figure 7-2 – Consolidation model

In the conversion model shown in Figure 7-3, a media server can be divided into two parts: a session control function and a media delivery function. These functions can be separated by an open interface and realized independently from each other.

The session control function may support all the protocols residing in every user device. It enables the media server to control the different sessions in a unified manner, regardless of the protocols running on user devices. It also manages the resource availability of media delivery function.

The media delivery function transmits the media content to the user device, under the control of session control function. The format converter in the media delivery function performs the key tasks of translation of the metadata information (e.g., resource identifier, resource size). Other key tasks are re-containerization, adjusting media content in the correct segment format. This means that the format converter transforms a certain metadata and segment format to another, appropriate, one. The format converter enables a media server to cache individual items of media content from a content source instead of all. Furthermore the separation of session control and media delivery functions can significantly increase their flexibility in selecting a proper protocol, metadata and stream formats.

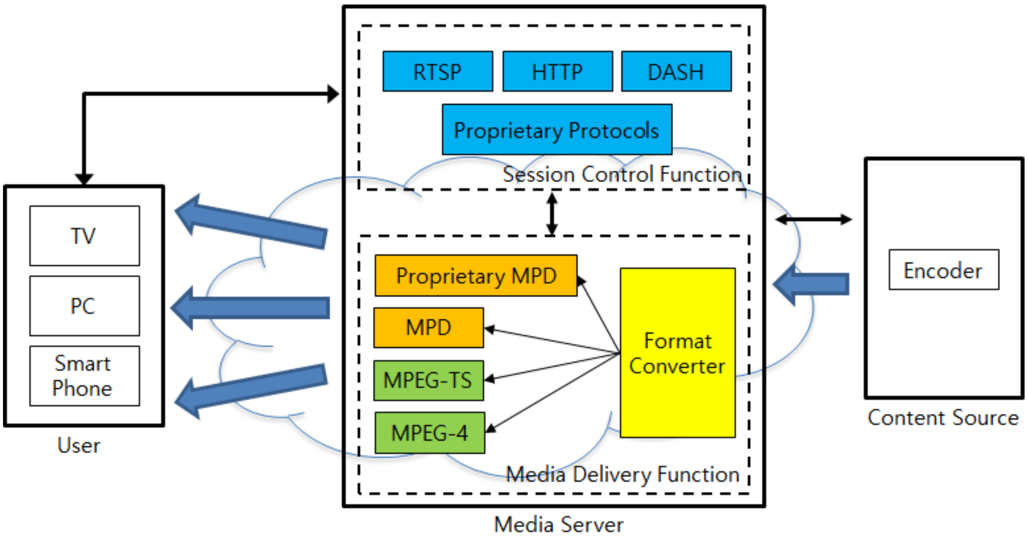

Figure 7-3 – Conversion model

8 Service requirements

This clause specifies service requirements of N-screen services when the conversion model is applied in a media server. The service requirements are specified as follows:

- It is recommended that the media server separate the session control function from the media delivery function.
- It is recommended that the media server select a particular session control function supporting the capabilities (e.g., metadata, stream format) that a user device handles.
- It is recommended that the media server service provide a capability to exchange information about the status of both network resources and user device resources.
- The media server is required to provide a capability to negotiate an appropriate quality of service (QoS), considering the status of both network resources and user device resources.
- The media server is required to select a proper media delivery function (e.g., metadata, stream format) byh examining request messages [e.g., hypertext transfer protocol (HTTP) request] or retrieving information (e.g., user preferences, user device capability).
- The media server is required to retrieve information (e.g., user preferences, user device capability) by communicating with a user profile server.
- It is recommended that the media server transmit media content to a variety of user devices with different media delivery capabilities.

- The media server is required to translate the metadata format to one appropriate to the user device.
- The media server is required to translate the stream format to one appropriate to the user device.

9 Security considerations

This Supplement is recognized as an enhancement of Internet protocol-based (IP-based) networks. Thus, it is assumed that security considerations in general are based on the security of IP-based networks and especially when required that they follow the security considerations identified in clauses 7 and 8 of [ITU-T Y.2701].

Appendix I

Content formats and delivery protocols

This appendix introduces a variety of content formats and delivery protocols and identifies their features by comparing them. Based on this, interworking components are clarified, and then resolved in terms of protocol, metadata and content format.

With the increasing variety of devices and their respective content formats, cross platforms like multiscreen service are increasingly necessary, providing that any content should be available on any device over any network connection at any time. Driven by the explosion in video-enabled devices like smartphones, new tablets, laptops, IP-based set-top boxes and now Internet-enabled television sets, consumers are moving to the IP-based N-screen service market. In addition to the growing amount of content itself, users are also able to access this content on a wider variety of devices both inside and outside the home.

The multiscreen service platform therefore can cache and deliver content in all the content formats required by a wider variety of devices and their respective content delivery protocols. This means that it can deliver any content to multiple devices requiring specific formats. The content would be OTT, VoD and linear TV programming. These devices can be connected to a kind of networks; 2G/3G/4G, WiFi, Ethernet.

Transmission of content between different nodes on a network can be performed in a variety of ways. Although various media streaming protocols available today differ in implementation details, we can classify them into two main categories: real-time transport protocol/real-time streaming protocol- (RTP/RTSP)- and HTTP-based streaming protocols. In this regard, some profiles can be identified in terms of content delivery protocols and content formats.

I.1 Media delivery protocol

(1) RTP-based streaming protocol

To provide a standard for mobile multimedia streaming, 3GPP defined the packet-switched streaming service (PSS) for unicast streaming of speech, audio, video and subtitles to 2G/3G/4G mobile terminals. 3GPP PSS is currently the most mature standard in this field, and all major mobile telecommunication equipment providers as well as streaming equipment providers support it. Therefore RTSP/RTP-based streaming protocol is introduced by referring to 3GPP PSS [b-3GPP TS 26.234].

3GPP PSS is mainly based on protocols developed by IETF. The main protocols include the RTSP for session control, the session description protocol (SDP) for presentation descriptions, and the RTP for media transport. In addition, HTTP is used for download of scene and presentation descriptions.

In RTSP/RTP-based streaming protocols, once a server and a device establish a connection, the server streams packets to the device until the device stops or interrupts the session. Consequently, the server maintains a session state with the device and listens for commands from the device regarding session-state changes. The RTSP, specified in [b-IETF RFC 2326], is used for session control.

The RTP was designed to define packet formats for audio and video content along with stream-session management, specified in [b-IETF RFC 3550]. The RTP usually runs on the user datagram protocol (UDP), a protocol without any inherent rate control mechanisms. This lets the server push packets to the device at a bitrate that depends on an application-level client/server implementation rather than the underlying transport protocol, and makes RTP a nice fit for low-latency and best-effort media transmission.

In conventional RTSP/RTP-based streaming, the server transmits content at the media encoding bitrate to match the media consumption rate of the device. In normal circumstances, this ensures that device buffer levels remain stable over time. It also optimizes the use of network resources because the device usually cannot consume at a rate above the encoding bitrate; consequently, transmitting above that rate would unnecessarily load the network. Moreover, transmission above the encoding bitrate might not even be possible for live streams in which the stream is encoded on the fly. However, if packet loss or transmission delays occur over the network, the packet retrieval rate of the device can drop below its consumption rate, which might drain its buffer and eventually result in a buffer underflow that interrupts the playback. This is where bitrate adaptation comes into play.

An adaptive streaming so that a server can dynamically switch to a proper bitrate by considering device status, contains two main phases, the setup phase and the media-streaming phase.

At session setup time, the device and server exchange information using UAProf, RTSP and SDP. This sets up the media streams and allows device and server to negotiate which features to use and how they are to be configured when used.

During session setup, the server sends a list of available streams and their properties such as bitrate and codec to the device via the SDP, specified in [b-IETF RFC 4566], in response to the RTSP DESCRIBE message of the device. After receiving the session description, the device selects from within the available audio and video streams that which best fits its link speed and decoding capabilities. The device then sends an RTSP SETUP message to the server to prepare it to send out the selected streams.

With the information in the SDP, the device can setup the media streams in the streaming session. In the SETUP and PLAY commands, the device includes three different RTSP headers. These allow the device to provide the server with guidance on the QoS parameters the link provides, the expected initial rate and its maximum available buffer size.

In response to the SETUP and PLAY messages, the server initiates the media-streaming phase. Bandwidth monitoring is usually performed on the device, which also computes network metrics such as round-trip time (RTT), packet loss and network jitter periodically. The device can use this information directly to make decisions about when to switch to a higher or a lower bitrate stream, or it can communicate this information along with its buffer levels to the server via receiver reports and let the server make those decisions. Such reports are usually transmitted via the real-time transport control protocol (RTCP).

Using the feedback provided in the RTSP and RTCP messages, a server can adapt the transmission rate and content rate to the current link conditions. A server needs the capability to adapt the average content rate to approximately the same bit rate as the transmission link so as to avoid buffer underflows.

(2) HTTP-based streaming protocol

With the increase of Internet bandwidth and the tremendous growth of the world wide web, the value of delivering audio or video data in small packets has diminished. Multimedia content can now be delivered efficiently in larger segments using HTTP. HTTP streaming therefore has become a popular approach in commercial deployments. However, each implementation uses different manifest and segment formats and therefore, to receive the content from each server, a device must support its corresponding proprietary device protocol.

In the meantime, standardization of the HTTP-based streaming protocol has also made great progress in various standards bodies including 3GPP [b-3GPP TS 26.247], MPEG, and OIPF [b-OIPF-HAS].

MPEG has recently released a first public draft of the new DASH specification. MPEG DASH is based on the 3GPP and partly also on the OIPF specifications. Therefore, the HTTP-based streaming protocol is introduced by referring to MPEG DASH, although there is a slight difference between them.

In the HTTP-based streaming protocol, the media device is the active entity that requests content from the media server. Therefore, server response depends on the requests of the device.

Consequently, the bit-rate at which the device receives the content depends on the device and available network bandwidth.

Similar to methods used in RTSP/RTP-based streaming, HTTP-based streaming protocol uses bitrate adaptation to prevent buffer underflow. The media content is divided into short-duration media segments (also called fragments), each of which is encoded at various bitrates and can be decoded independently. When the device plays the fragments back to back, it can seamlessly reconstruct the original media stream. During download, the media device dynamically picks the fragment with the right encoding bitrate that matches or is below the available bandwidth and requests that fragment from the server. This way, the device can adapt its media-consumption rate according to the available receive bandwidth.

I.2 Metadata

1) Session description protocol

   The SDP provides a standard representation to convey media details, transport addresses and other session description metadata to participants. It is purely a format for session description so that it does not incorporate a transport protocol. The SDP is intended to describe multimedia sessions for the purposes of session announcement, session invitation and other forms of multimedia session initiation. For example, an RTSP device and server negotiate an appropriate set of parameters for media delivery, using SDP syntax to describe those parameters.

2) Media presentation description

   Dynamic HTTP streaming requires various bitrate alternatives of the multimedia content to be available at the server. In addition, the multimedia content might consist of several media components (e.g., audio, video and text), each of which might have different characteristics. In MPEG-DASH, these characteristics are described by a media presentation description (MPD), which is an extensible markup language (XML) document.

I.3 Stream format

The multimedia content can be accessed as a collection of segments. A segment is defined as the entity body of the response to the HTTP GET or a partial HTTP GET of the DASH device. A media component is encoded and divided into multiple segments. The first segment might be an initialization segment containing the required information for initialization of the media decoder of the DASH device.

MPEG-DASH defines segment-container formats for both ISO base media file format and MPEG-2 transport streams. On the other hand, the 3GPP file format (3GP) was introduced in 3GPP Release 5 and is based on the ISO base file format structure. It extended the ISP base file format to support the PSS codecs and other 3GPP specific features. Capability exchange was introduced based on the user agent profile (UAProf) protocol to allow servers to match content to the device rendering capabilities [b-3GPP TS 26.247].

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