Request for Technology of the JT-NM

12 September 2013

This is the **Request for Technology** (RFT) of the *EBU/SMPTE/VSF Joint Task Force on Networked Media*.

A meeting on 14 September will take place in Amsterdam to answer questions. Please RSVP to bob.ruhl1@verizon.net.

For more info on the *Task Force*, go to tech.ebu.ch/jt-nm.

For any question, please e-mail to jt-nm-rft@videoservicesforum.org.

Ownership and copyright

Request for Technology

EBU/SMPTE/VSF
Joint Task Force on
Networked Media (JT-NM)

Executive summary

The European Broadcasting Union (EBU), the Society of Motion Picture and Television Engineers (SMPTE), and the Video Services Forum (VSF) are co-publishing this Request For Technology (RFT) as part of the activities of the *Joint Task Force on Networked Media (JT-NM)*. The *Joint Task Force on Networked Media* [hereafter referred to as "we" and the *Task Force*] has been created to help manage the transition from infrastructures that is based on speciality broadcast equipment and interfaces (SDI, AES, etc.) to IT infrastructure and packet networks (Ethernet, IP, etc.) This effort spans the entire professional media industry and all of its applications, including live and file-based. The *Task Force* is an open initiative, and we invite you to join.

The *Task Force* has already collected business-driven user requirements and published the Report on User Requirements. This RFT is now released in order to identify the technologies, current or in development, that can fulfil some requirements. Our next step will be to conduct a gap analysis between the user requirements and the responses to this RFT. Finally, a report of the results will be published by November 30, 2013. During the Gap Analysis, we will look at each response and note which user requirements the respondent claims they satisfy. We will aggregate these responses and then report on any user requirements where no respondent submitted a technology that addresses the requirement. The *Task Force* is not doing any "shootouts" or other comparative analyses to determine a so called "winner".

Future activities will depend upon submissions received in response to this RFT and the subsequent Gap Analysis. It is important to note that there may be a mix of follow-on activities that are carried out by the *Task Force*, and there may be other activities that are carried out by individual organizations or other industry groups. For example, we may recommend subjects for standardization.

This RFT is being released now because already, several proprietary networked media solutions exist. On the other hand, there is a demand in the industry for interoperable, open systems that allow the mixing and matching of products from different vendors to meet users' needs. There is a strong sentiment both in the user and manufacturer communities that managing the transition from SDI/Audio streams to this new infrastructure is critical in order to provide the required user functionality and to avoid waste both in terms of cost and time.

Respondents to this RFT may submit one or many Technologies that address at least one Use Case.

All interested parties are invited to respond to this RFT. An intent to respond must be received by 11 October, 2013 and deadline to respond by 1 November, 2013. Respondents need not be a member of any of the sponsoring organizations. All Technologies submitted to the RFT shall be accompanied by an IPR Declaration.

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Part 1 - Scope & Objectives

1. Introduction

This first part covers the scope and the objectives of this RFT and provides background on the *Task Force*.

The second part of this RFT covers Submission Guidelines and the Third part defines the Technologies and User Requirements together with the information elements that are required in any submission.

2. What problem are we trying to solve?

The Joint *Task Force* on Networked Media has been created to help with managing the transition from broadcast infrastructures that are based on speciality broadcast equipment and interfaces (e.g, SDI, AES3/AES10, Synchronisation, control data elements, etc.) to infrastructures based on IT and packet networks (e.g, Ethernet, IP, etc.). This effort spans the entire professional media industry and all of its applications, including live and file-based. (see §6).

3. How do we plan to solve this problem?

The *Task Force* started by gathering the business-driven requirements from professional media users. We now need to know if and how current and upcoming technologies can fulfil those requirements and what are the gaps that need to be filled in order to enable this transition.

To do so, this RFT is now asking Technology owners to respond in order to be included in the Gap Analysis. The results of the Gap Analysis will be published in a report.

Future activities will depend upon Responses to this RFT and the subsequent Gap Analysis. It is important to note that there may be a mix of follow-on activities that are carried out by the *Task Force*, and there may be other activities that are carried out by individual organizations or other industry groups. For example, we may recommend subjects for standardization.

4. What information are we seeking with this RFT?

The *Task Force* wants to identify the technologies, current or in development, that can fulfil a number of the User Requirements that have been identified. These Technologies can be a software or a hardware design, a protocol, a standard or a framework, a methodology or a process, or an ensemble of them.

For each of these Technologies, we are interested in knowing:

1. what are the specific User Requirements that are fulfilled,
2. how it can fulfil them, and
3. with what technical specifications.

We also need to know if these Technologies are available or implementable now and if not, when they will become so. Finally, we need to know about the IPR status of the Technology (see §13).

5. What are the expected benefits and why now?

This RFT is being released now because several proprietary networked media solutions already exist. On the other hand, ***there is a demand within the industry for interoperable, open systems that allow the mixing and matching of products from different vendors to meet users' needs***.

There is a strong sentiment both in the user and manufacturer communities that managing the transition from SDI (and other Media-Associated Data Payloads and associated links) to this new infrastructure is ***critical in order to provide the required user functionality and to avoid waste both in terms of cost and time***.

6. About the Joint *Task Force* on Networked Media

6.1 Who initiated this Task Force?

The European Broadcasting Union (EBU), the Society of Motion Picture and Television Engineers (SMPTE), and the Video Services Forum (VSF) are co-sponsoring the *Joint Task Force on Networked Media* (JT-NM). This fosters collaboration at an international and an industry-wide scale.

Annex C present the *Task Force* Vision and Mission and the Timeline it has identified for its work.

6.2 Who can join the Task Force and how?

The *Task Force* is an open initiative, and we invite all parties with a professional interest in the work of the *Task Force* to join in the work. Those wishing to participate should send an e-mail to Bob Ruhl, VSF Operations Manager (bob.ruhl1@verizon.net) and include their name, e-mail address, work affiliation and their interest in joining the *Task Force*.

Part 2 - Submission Guidelines

7. Introduction

This part provides the information that is needed to respond properly to this RFT. We invite you to read it carefully.

8. Respondents

All interested parties are invited to respond to this RFT. Respondents do not have to be member of the JT-NM, EBU, SMPTE nor VSF to respond.

9. Communications

All communications regarding this RFT should be directed to the *Task Force*'s RFT Management Team at jt-nm-rft@videoservicesforum.org. Please do not forget to indicate your Reference Number (if you refer to a Response, see §11.3), contact details and e-mail address when communicating with the RFT Management Team.

9.1 Single Point of Contact Required

Respondents shall provide a Single Point of Contact for all communications regarding the RFT. It is the responsibility of the Point of Contact to disseminate communications from the RFT management team appropriately within his/her organization.

9.2 Intent to Respond

We ask that you notify us by 11 October 2013, if you intend to respond. Such notification should be by e-mail to jt-nm-rft@videoservicesforum.org and should include the organization name and the Single Point of Contact.

9.3 Withdrawal of Responses

If you need to withdraw a previously submitted Response to this RFT, you must do this in an e-mail sent to jt-nm-rft@videoservicesforum.org before the cut-off date (see §10). You should receive a confirmation e-mail from the RFT Management Team acknowledging your withdrawal. If you do not receive a confirmation e-mail within 48 hours, you should send an e-mail to the RFT Management Team requesting acknowledgment of receipt of your withdrawal.

9.4 Respondent Meeting

The RFT Management Team will hold an online meeting with Respondents on 14 October 2013, at which time the team will discuss the RFT and address any questions that Respondents may have. Respondents should be aware that in the interest of fairness, all questions and answer may be documented and may be shared with other Respondents, whether they attend the meeting or not.

9.5 Queries

It is recognized that when Respondents review this RFT they might need to contact the RFT Management Team at jt-nm-rft@videoservicesforum.org with points for clarification. The team will attempt to assist Respondents with background information, additional explanations, etc. Respondents should be aware that, in the interest of fairness, all questions and answers which serve to clarify the RFT or which provide additional information may be shared with other Respondents.

9.6 Sharing the RFT

This RFT is a public document. Provided that it is not modified in any way it may be passed on to other parties who may have a bona-fide interest in responding to the RFT.

9.7 Partial Responses

We understand that it is unlikely that any single technological solution will fully address all the User Requirements described in this RFT. Partial responses are perfectly acceptable.

9.8 Less is More

In providing the required information, you can always provide additional details or refer to complementary documents that you can include in your submission. Be aware that considering the very aggressive timeline of the Gap Analysis that will follow, it may be in the interest of Respondents to highlight the most essential information to be considered.

10. Dates

The anticipated time schedule is as follows.

- Publication of this RFT on 12 September 2013 for the IBC.
- RFT Presentation and Q&A at the IBC in Amsterdam on 14 September 2013.
- Intent to respond must be received by Management Team no later than 11 October 2013 at 23.59 EDT.
- Online Q&A meeting with Respondents on the 14 October 2013.
- Questions about the RFT sent in via e-mail will receive written responses or will be subject to teleconferences and/or web meetings until 25 October 2013.
- Cut-off: Responses to the RFT must be received by the Management Team not later than 1 November 2013 at 23.59 EDT.
- Publication of Gap Analysis results on 30 November 2013

11. Submission Procedure

11.1 Submission Format

Respondents are advised to use the RFT Submission Template provided in Annex B. This is a template that provides a structure to submit the requested information. Although the Respondent need not strictly follow the template, all the information requested in the template must be covered and should be set out in the order provided in the template. This will be extremely helpful during our Gap Analysis process. Responses will only be accepted in electronic form; paper submissions are not allowed.

11.2 File naming and numbering

- Files should follow this naming convention: JTNMxxx-1.yyy where xxx is your submission reference number and yyy is the common file suffix for the document file format.
- If you are uploading multiple documents, please use the ‘zip’ format to compress them into one file.
- If you need to revise a document, please increment the last number in the file name.

11.3 Submission Steps

1. Respondents should contact the RFT team at jt-nm-rft@videoservicesforum.org as described in §9.2, above, to indicate their intent to respond. They will obtain a confirmation with a submission Reference Number. Respondents must include their Reference Number with their submission and in all correspondence regarding their submission.

2. When ready, please upload your submission to the location provided when you received your submission Reference Number.

3. Announce your submission by sending an e-mail to jt-nm-rft@videoservicesforum.org. You should receive a confirmation e-mail from the RFT Management Team acknowledging your submission. If you do not receive a confirmation e-mail within 48 hours, you should send an e-mail to the RFT Management Team requesting acknowledgment of receipt of your submission.

NOTE: To be considered as valid all Responses to this RFT must be submitted per the process described in this RFT. Verbal or written submissions which are not made per the submission guidelines described here will NOT be accepted.

12. Evaluation and Dissemination

12.1 Gap Analysis Report

During the Gap Analysis we will look at each Response and note which User Requirements the Respondent claims they satisfy. We will aggregate these responses and then report on any User Requirements where no Respondent proposed a Technology that addresses the Requirement. The *Task Force* is not conducting any "shootouts" or other comparative analyses to determine a so called "winner".

12.2 Rejection, Combination and Expansion of Responses

We may reject a submission if the Respondent fails to meet the submission criteria - for example, if a technology is submitted without an IPR declaration. There will not be any technical merit evaluations performed on submissions nor on the particular value of any particular submission as part of the Gap Analysis process. We may choose to accept portions of a Response, and we may merge concepts presented in multiple Responses into a single final report. We may also choose to expand on concepts presented in a Response to this RFT.

12.3 Visibility of Submissions

Respondents to the RFT accept and acknowledge that their contribution may be made public and that portions of Responses may be selected for inclusion in public reports and specifications. Working Drafts of our documents may also be made available to the general public.

13. Intellectual Property Rights

13.1 IPR Declarations required with Responses

All Technologies submitted in a Response to the RFT shall be accompanied by an IPR Declaration. Respondents must declare any intellectual property (including patents and patent applications) believed to be essential to the implementation of each of the Technologies submitted, whether or not owned or controlled by the Respondent.

The declaration is made upon the personal knowledge of the individual making the declaration. Nothing in this RFT requires a patent search on the part of any Respondent, nor does it require a Respondent to conduct exhaustive research into the IPR status of technology which is included in their submission. However, regarding technologies not owned or controlled by the Respondent, the Respondent shall notify the *Task Force* where the person making the submission is personally aware of intellectual property believed to be essential. For example, if MPEG-2 is included in a Response, the Respondent shall note that he or she is aware that there are numerous IPR claims associated with this technology. The *Task Force* is collecting these declarations for information purposes only. We intend to make declarations public.

Where such intellectual property is owned or controlled by the Respondent, the submission of each Technology must be accompanied by one of the following four Licensing statement:

No License Required	The Respondent declares that no license is required to implement a given Technology submitted in their RFT Response.
Compensation-Free, Reasonable and Non-Discriminatory License (RAND-Z)	The Respondent declares that they will grant a license to all implementers regarding a given Technology submitted in their RFT Response without a requirement for monetary compensation (i.e. no royalty or other fee).
Reasonable and Non-Discriminatory License (RAND)	The Respondent declares that they will grant a license to all implementers regarding a given Technology submitted in their RFT Response except that such Respondent may charge a reasonable and non-discriminatory royalty.
Unwilling to Commit to Any of the Above Options	The Respondent is unwilling to commit to any of the above license declaration statements.

Submissions which do not contain an IPR Declaration will be rejected.

13.2 Standardization IPR policies

This RFT and Gap Analysis process may trigger further standardization activity based on Technologies that were submitted.

To be considered for standardization, a Technology must comply with the IPR policy of the standards body under which the standard is published. For instance, an important standards body for the broadcast industry is the SMPTE. You can review its policy by downloading: https://www.smpte.org/sites/default/files/SMPTE_IP_Policy_2013-08.pdf

Part 3 - Technologies and Requirements

14. Introduction

This part defines what is a Technology for the purposes of this RFT and explains the elements of information that are requested in this RFT.

15. Definitions

The following are definitions of terms that are used in this RFT.

- **COTS**: Commercial Off-The-Shelf is a term defining common commercial products that are widely available and useful by the IT industry. The products may be "enterprise worthy" or "garden variety" depending on user requirements. Many user stories request COTS equipment. However, non-COTS technology is permitted if necessary.
- **Media-Associated Data Payload**: Data payloads are the data elements carried by the physical links. These payloads are defined as media/essence or related. For this definition a transport link is composed of two strata; lower layers composed of physical, modulation formats, framing and packet aspects and higher levels composed of at least data payloads. A Media-Associated Data Payload is comprised of (one, some or all) the following data types; SDI payloads, AES3 audio payloads, MADI/AES10 payloads, timed text (closed captions) payloads, metadata elements, XML/JSON elements, control data elements and other/future TBD payloads. Data may be packaged on a link as (one, some or all); synchronized collections, multiplexed collections, or as separate independent elements.
- **Stream**: A stream is an ordered sequence of bits sent from a destination to a receiver. For media streams, the timing is normally "real-time" or a multiple of or fraction of a real time value. Streams can be pulled or pushed from a source to one or more destinations. SDI is an example of a pushed media stream technology.
- **File**: A file is a data record. It is typically stored for later access. It may be created on demand. It is often transferred using networked links. Media file transfers are typically sent at slower or faster than real-time media rates. Files can be transferred as full, partial sections or out-of-order sections depending on workflow needs. Files can be pulled or pushed from a source to one or more destinations.

16. Technology Submitted

In the context of this RFT, Respondents can submit one or many Technologies that address at least one Use Case that is in the scope of this RFT. The Technologies that are submitted do not need to be able to satisfy every User Requirement, and partial responses are acceptable.

Each Technology that is submitted needs to be described separately. A Technology description includes at least a name, a high-level description, what Type of technology (see §17) and information about the availability of the Technology and an IPR statement concerning the Technology.

Responses which recognize that networked media will exist alongside traditional SDI media infrastructures (and other Media-Associated Data Payloads and associated links) and that describe the bridge between these infrastructures are also encouraged. Respondents are welcome to present Technologies that are not an exact replacement for SDI - after all, SDI already exists. One can envision any number of innovative capabilities that could be enabled if the strict timing and delay characteristics of SDI were relaxed. In other words, if you would like to propose an SDI replacement, that would be great. If you foresee some new and innovative capabilities that can be enabled by networked media, and one or more of our Use Cases touch on it, then please propose it.

17. Types of Technology

A Technology, as defined in this RFT, can be any technological solution that addresses at least one Use Case. It can be, but is not limited to, a software or hardware design, a protocol, a standard, a framework, a methodology or a process, or an ensemble of these. We are not looking for specific products but references to products can be helpful as information to illustrate your submitted Technology.

The Technologies that can be submitted will generally fall into one of four Types:

1. **Grand solution sets**. This type would cover all or large portions of the functionalities needed to implement most Use Cases. For example, a submission could be an ensemble of solutions covering 80% of all Use Cases.

2. **Point solution**. This type would cover the functionalities needed to implement one or more Use Cases. For example, a submission could cover route pinning which would increase predictability in network performance and improve Quality of Service.

3. **Pure technology for reuse**. This type would provide specific technologies to be used in creating solutions to implement one or more Requirements. For example, a submission could cover how to map an SDI-payload over IP.

4. **Configurations for COTS equipment**. This type would define configuration settings for appropriate Commercial Off-The-Shelf (COTS) equipment to implement specific User Requirements. For example, a submission could cover how to use and configure an IEEE Standard switch protocol to route media packets with a specified QoS.

The Response should state which of a these four Types best fits each Technology that is submitted.

18. Use Cases and User Requirements

All the User Requirements for this RFT are grouped into sixteen Use Cases. Each Use Case groups many requirements that are put in the context of who needs them and what is the business value of having them fulfilled. The full list of Use Cases and their Requirements is in Annex A.

For each Technology that is submitted, one or many Use Cases need to be addressed. The Response shall state which User Requirement(s) are fulfilled by each Technology that is proposed.

For each of these Use Cases, it must be indicated if the Requirements are fulfilled “Fully”, “Partially” or “Not”.

Fully	All the aspects of the Requirements are fulfilled by this Technology
Partially	Some of the aspects of the Requirements are fulfilled by this Technology
Not	None of the aspects of the Requirements are fulfilled by this Technology

If “Fully” or “Partially” is to be cited, a summary explanation of how this Technology can fulfill the Requirement must be given. If “partially”, describe the aspects of the Requirements are not fulfilled. If relevant, technical specifications can be provided to support the explanation as well as reference to a separate document.

Annex A - Detailed Use Cases

The JT-NM collected 136 unique user stories from media organizations, manufacturers and consultants that identified a number of User Requirements for networked professional media. This annex grouped these stories into sixteen Use Cases which are composites attempting to capture the overall spirit of the original stories.

For each Technology that is submitted, one or many Use Cases need to be addressed. The Response shall state which User Requirement(s) are fulfilled by each Technology that is submitted.

It should be kept in mind that the order of these Use Cases and the order of the User Requirements in them does not reflect any prioritization. See the definitions at §15 for clarity of terms.

> *NOTE: Additionally, Respondents may reference any of the original User Story listed in the Task Force publication Report on User Requirements, which is available at http://tech.ebu.ch/jt-nm, if necessary.*

A1. Configuration (CONFIG)

As a facility operator, I want to have flexible error-free configuration to:

(CONFIG-1) be able to quickly add and configure new equipment and elements;
(CONFIG-2) be able to auto-discover devices attached to the network;
(CONFIG-3) be able to have the configuration of devices be intelligent and highly automated;
(CONFIG-4) be able to have an excellent management/monitoring view of the system;
(CONFIG-5) be able to deal with the variety of formats, stream-types, and file types.

So that I can be on-air quickly, avoid the human mistakes and errors associated with high complexity repetitive engineering tasks, to understand faults in a timely manner.

A2. Commercial Off-The-Shelf (COTS)

As a systems designer I would like to deploy commercial IT technology for use in professional media applications to:

(COTS-1) take advantage of the marketplace economics of IT technology including packet-based networking, servers and storage;
(COTS-2) make use of the extensive and well-trained base of design, operations, and maintenance personnel available in this field;
(COTS-3) deploy enterprise-class capabilities and redundancy options;
(COTS-4) use any one of a number of monitoring, diagnostic and troubleshooting tools that currently exist for enterprise deployments of IT infrastructure.

So that I can reduce the total cost of ownership of my professional media operations.

A3. File-based (FILE)

As a facility or production company owner, a producer or content provider, or a system engineer, I want to:

(FILE-1) be able to mix streaming-based and file-based content in the same unified packet-based system that conforms with published standardized specifications;

(FILE-2) be able to begin work on “post-production” on live content as it is being captured;
(FILE-3) be able to view what the program will look like in near real time;
(FILE-4) be able to transcode, analyze and transform content on-the-fly.

So that I can shorten the production cycle and meet the needs of the downstream consumers of media.

As a video editor, I want to:

(FILE-5) be able to mix media of various qualities (codecs, data rates, etc.);
(FILE-6) be able to change dynamically between streaming and high-quality transfers.

So that I can get the best signal and content quality while editing on low-bandwidth connections.

A4. Formats (FORM)

As a participant in the television equipment ecosystem (such as a vendor, integrator, architect or operator), I want the signal formats inside the packet-based media networks of the future television plant to:

(FORM-1) be well documented through the use of open and interoperable standards;
(FORM-2) be supportive of current media processing operations such as mixing, cross-fading, DVE, and voiceover;
(FORM-3) be compressed or uncompressed, with configurable sub-sampling and sample bit depth;
(FORM-4) if compressed, to be able to support arbitrarily good quality (up to lossless if desired) even with multiple compression concatenations of a typical chain through a broadcast plant;
(FORM-5) be based on well-understood and generally-available compression and networking technologies;
(FORM-6) be able to address parts of signals (audio, video, metadata) in addition to whole signals;
(FORM-7) be able to support current and future image formats, frame rates, and file types;
(FORM-8) support the ancillary streams needed by some of our viewers and/or required by regulatory agencies to be carried such as Closed Captions, subtitles, audio description, and multiple languages;
(FORM-9) to allow addressing of arbitrary data events, including those synchronised with content signals;
(FORM-10) be able to flexibly deploy and interactively control both software- and hardware-based real-time signal processing and analysis modules for packet-based flows.

So that high-functionality facilities can be constructed using equipment from multiple vendors with an expectation of excellent interoperability and a high-quality output signal.

A5. Interoperability (INTEROP)

As a system architect, product designer, manufacturer or content provider, I want to:

(INTEROP-1) be able to use readily available and accepted packet-based standards, technology (e.g., IEEE and IETF standards for networking), interfaces (e.g., APIs), components and products in a multivendor environment;
(INTEROP-2) be able to ensure that all network-attached devices are designed and tested to operate in likely real-world scenarios;
(INTEROP-3) be able to ensure that all network-attached devices are able to appropriately handle dropped packets and out-of-order packet delivery;

(INTEROP-4) be able to have control surfaces that are conceptually decoupled from the software control APIs of the underlying infrastructure and equipment;

(INTEROP-5) be able to design and manufacture systems and test compliance to an industry-standard interoperability specification;

(INTEROP-6) be able to interoperate with key existing media, synchronization, and metadata protocols (such as, for example, SDI, AES audio, SMPTE 12M, SMPTE ST-2022 series, SMPTE RDD-6, SCTE 35);

(INTEROP-7) be able to use IPv4 or IPv6 (for an IP-based solution);

(INTEROP-8) be able to store, retrieve and exchange media and information between media production systems using media production-oriented standards-based protocols.

(INTEROP-9) be able to use “self-contained” / “self-defining” streams with software-defined connections and/or physical-only connections;

(INTEROP-10) be able to include communications (e.g., “intercom”) along with content streams;

So that my operations are optimized, I can have maximum vendor sourcing flexibility through “plug-and-play”, “future proof” my system designs, I can choose the appropriate human interfaces for the evolving workflows independently of core infrastructure, maintain quality and compliance with broadcast regulations (e.g., US FCC CALM), I can manage the large (and growing) number of network-attached device addresses, and I can meet the media format needs of my downstream customers.

A6. Monetization and Revenues (MONETIZE)

As a professional media content producer, I want to:

(MONETIZE-1) distribute content to end users and to content aggregators over public packet-based networks, with clear traceability and rights management;

(MONETIZE-2) be able to adapt content and advertisements to end user in real-time based on their feedback and information;

(MONETIZE-3) allow the viewer to compose the audio/video, pull contextual data and interact with me lively;

(MONETIZE-4) monitor media resources (network/processing/storage) usage.

So that I can gain more revenue from each of my content sources, through larger numbers of subscribers, maximize benefits for us getting better advertiser’s satisfaction and personalized user experience and I can bill to service usage.

A7. Provisioning (PROV)

As the systems engineer of a professional media facility I want to:

(PROV-1) be able to use state-of-the-art tools to deploy professional media connectivity whenever and wherever I need it;

(PROV-2) be able to send professional content over the Internet, meeting our quality needs, but taking advantage of the self-routing and self-provisioning capabilities of the Internet;

(PROV-3) be able to rapidly (and in some cases, automatically) set up streams from new devices;

(PROV-4) be able to have my infrastructure scale automatically with load balancing capabilities that take advantage of various links available;

(PROV-5) be able to have my workflow automatically adjust to incorporate the correct transcoding so that when I provision a stream, the format type at the destination node is correct;

(PROV-6) be able to quickly set up efficient distribution networks that deliver the same content to multiple places;

(PROV-7) be able to provision a link at a low quality initially, if that is all that is available, but then allow the quality to improve as resources become available.

So that I can rapidly meet the business-driven operational needs of my company and make economical decisions about the links I use for transport of professional media.

A8. Quality of Service for File Transport (QOS-FT)

As a system designer or facility operator I want to transport media files between endpoints in non-real-time using a packet-based network with:

(QOS-FT-1) adjustable and deterministic transfer time, including faster-than-real-time if desired;

(QOS-FT-2) upper-end bounded data loss; (define a max transport loss %)

(QOS-FT-3) rate-sufficient to meet the needs of current and future format payloads;

(QOS-FT-4) transport over local, campus networks and Internet;

(QOS-FT-5) multiple defined QoS levels for file transfer based on job, workflow, source or destination;

(QOS-FT-6) the ability to monitor QoS deliver-to-commit and to make adjustments by priority criteria;

(QOS-FT-7) profiles of service to support a variety of workflows. One goal is to provide deterministic file transfers with a known transfer time. For example,

a. Class A: superior QoS similar to what a lossless, high bandwidth, low latency LAN can provide today.
b. Class B: relaxed Class A profile. One or more parameters are relaxed to create a “good enough” profile for many real world use cases.
c. Other classes if needed.

So that I can configure agile file-based media workflows and transport media files using the packet-based network in my facility, be able to select between QoS profiles and trade off costs and performance depending on business needs, and to ensure that files are consistently delivered when they are needed.

A9. Quality of Service for Streams (QOS-S)

As a system designer or facility operator I want to transport synchronized, end-to-end, real-time, muxed or individual, audio/video/metadata streams over the packet-based network with:

(QOS-S-1) video-frame/audio-sample time accuracy (see Timing case);

(QOS-S-2) very low latency;

(QOS-S-3) lossless transport;

(QOS-S-4) a rate sufficient to meet the needs of current and future format payloads;

(QOS-S-5) transport over local and campus networks;

(QOS-S-6) each stream or group of streams having selectable QoS profile that is defined by the system configuration;

(QOS-S-7) profiles of service to support a variety of workflows. For example,

a. Class A: superior QoS similar to what the SDI ecosystem provides today. This is a “near SDI” profile but not equivalent in every aspect. This also applies to Media-Associated Data Payloads and their links, not just SDI.
b. Class B: relaxed Class A profile. One or more parameters are relaxed to create a “good enough” profile for many real world use cases that do not require the full feature set of SDI, for example.
c. Other classes if needed.

So that I can configure agile media workflows and transport real-time AV streams using the packet-based network in my facility and be able to select QoS profiles and tradeoff costs and performance depending on business needs.

A10. Reach (REACH)

I want to exploit the near-ubiquitous reach and rapidly increasing bandwidth of the globally connected packet-based networks (including private leased links and also the public internet) in order to:

(REACH-1) be able to easily, securely, effectively browse media and exchange files with peers at other organizations;
(REACH-2) be able to quickly create ad-hoc live interconnections that are able to utilize the available network;
(REACH-3) be able to combine the above to leverage geographically distributed content, staff, and equipment as if they were inside my four walls;

So that I can improve time-to-air and improve staff, equipment, and budget utilization.

A11. Reliability (REL)

As a professional media organization, I want to:

(REL-1) implement redundant paths in my network to ensure that the facility does not contain single points of failure;
(REL-2) identify primary and backup paths of the same stream; redundancy switching among those paths should be seamless;
(REL-3) ensure that a failure of one system in a studio is contained within that system and cannot affect other systems in that studio, or other studios in that facility;
(REL-4) eliminate making on-air mistakes;
(REL-5) include an equivalent function of the broadcast “tally” system in the packet-based network so that devices downstream or, in a routing infrastructure, can understand a bidirectional (upstream/downstream and vice-versa) status of “on-air” so that inadvertent system changes could be locked-out (or prioritized to administrational / override) status;
(REL-6) know the key system reliability specifications that constitute "enterprise-class" network equipment that will be able to transport high-bitrate video signals in a live television production environment.

So that broadcasting can continue without interruption even in the event of failures (including configuration errors) of shared systems, so that I can recover from a link failure without having time gaps in the media, and so that I can effectively communicate with suppliers to explain my requirements and appropriately evaluate products for use in my facility.

A12. Security (SEC)

As a broadcast media organization, I want to:

(SEC-1) protect against unauthorized access from within the organization or from outside the organization to data, systems control, or media;
(SEC-2) protect against attacks that disrupt the proper function of the organization;
(SEC-3) have appropriate administrative control systems to support dynamic access control to organization systems;
(SEC-4) have appropriate security monitoring and alarming.

So that restricted or sensitive material does not leak to unauthorized users, I can prevent my operation from being disturbed by malicious actions and no one can conduct unauthorized activities under the name of my organization.

A13. Streams (STREAM)

As a system designer or facility operator I want facility-wide media/data real-time streaming so I can stream:

(STREAM-1) real time audio, video, ancillary data and metadata that can be synchronized and/or multiplexed together or sent separately (see Timing case).
(STREAM-2) self-describing streams that can carry identifiers such as stream unique identifier, stream name, stream contents, and stream content owners;
(STREAM-3) virtual bundles: separate streams and data paths logically grouped as one;
(STREAM-4) nearly equivalent to SDI or other Media-Associated Data Payloads and their associated links in terms of transport functionality (see Quality of Service for Streams case);
(STREAM-5) across an infrastructure enabled to carry future payloads (such as UHDTV);
(STREAM-6) in a point-to-point or point-to-multipoint fashion as desired;
(STREAM-7) such that media is switchable on video or audio frame boundary (see Timing case);
(STREAM-8) across an infrastructure that scales from small to large installations;
(STREAM-9) between any nodes connected to the packet-based network;
(STREAM-10) and be able to use software-based real-time signal processing and analysis of streams;

So that I can build agile, real time, lossless, low latency, workflows with the ability to trade off QoS, formats, and reach.

As a video editor, I want to:

(STREAM-11) be able to mix media of various qualities (codecs, data rates, etc.);
(STREAM-12) be able to change dynamically between streaming and high-quality transfers;

So that I can get the best signal and content quality while editing on low-bandwidth connections.

A14. Sustainability (SUST)

As a professional media organization, I want to:

(SUST-1) be able to separate the physical locations of control surfaces, displays, video and network processing gear to the most appropriate locations for energy usage, efficient cooling, and noise;
(SUST-2) reduce the weight and size of broadcast equipment to be deployed in the field through aggregating multiple streams on a single physical layer connection;
(SUST-3) monitor resources (network/processing/storage) usage;
(SUST-4) minimize the energy consumption of storing, streaming and moving media around the network, particularly when idle;
(SUST-5) be able to easily repair, upgrade, maintain and disassemble the equipment when decommissioned;
(SUST-6) ensure the longevity of my design by using future proof technologies;

So that I have the freedom to deploy people and technology in the most cost and process efficient way, save on transport cost, installation time and travelling of operating staff, pay only for the resources that I use, I can also meet “carbon consumption” regulations, reduce OpEx on energy spend and carbon tax, and protect myself against possible future resource shortages.

A15. Test & Monitoring (TESTMON)

As a facility owner, a media system reseller, a maintenance person, a network operator or an administrator I want to:

(TESTMON-1) be able to simply identify streams;

(TESTMON-2) be able to monitor full-quality stream audio, video, and metadata at any point in the facility by multiple simultaneous users;

(TESTMON-3) be able to monitor thumbnail views of any video stream, with audio bars and other metadata displayed;

(TESTMON-4) be able to view exception-based monitoring alerts of any stream (such as presence of video/audio/captions) and set off audible alarms based on these;

(TESTMON-5) be able to quality test streams including pass/fail non-destructively in a straightforward manner;

(TESTMON-6) be able to test encrypted and non-encrypted streams;

(TESTMON-7) be able test correctness of compressed bitstreams;

(TESTMON-8) be able to test streams for standard broadcast-style quality measures and standards and for packet-based quality measures and standards;

(TESTMON-9) be able to verify compliance of the end-to-end packet-based network infrastructure to specifications for installation, function, performance, reliability and interoperability;

(TESTMON-10) be able to monitor media network traffic;

(TESTMON-11) be able to monitor systems for compliance with QoS/SLA agreements or for system commissioning and acceptance;

(TESTMON-12) be able to observe packet-based network statistics and trends;

(TESTMON-13) be able to decouple monitoring from mechanism used for media stream transport content for reliability;

(TESTMON-14) be able to see a ‘dashboard-view’ roll-up of important routes and flows in my facility;

(TESTMON-15) be able to remotely monitor all system parameters in real time;

(TESTMON-16) have a consistent amount of delay between the time a signal is present at the source and the time it appears at a monitoring point;

So that I can ensure that these complex systems are operating as required, diagnose, support and manage to QoS agreements, minimize overall costs and downtime, provide the Quality of Experience (QoE) that my consumers expect, quickly determine the location of errors or outages and take appropriate remedial action, and so that I can quickly and simply verify the presence or absence of critical systems to be able to troubleshoot and restore media services.

A16. Timing (TIME)

As a system designer I want facility-wide timing methods such that I can accomplish the following:

(TIME-1) keep multiple audio, video and data streams in the same transport in sync (lip sync);

(TIME-2) keep multiple media streams synced together (link sync);

(TIME-3) keep streams and end points synced to a common timing reference where required (nodal sync);

(TIME-4) enable frame (or audio sample) accurate switching of real time AV synced streams (synced switching);

(TIME-5) maintain phase-sync between audio streams (of a stereo/surround audio stream group)

So that I can coordinate facility streams in lock step for sourcing, sinking, mixing, displaying and grooming to create agile real time workflows.

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Annex B - RFT Submission Template

This is a template that provides a structure to submit the requested information. Although the Respondent doesn't have to follow it strictly, at least all this information must be covered and in this order to facilitate the analysis process. The Responses will only be accepted in electronic form; paper submissions will not be accepted.

Identification of Respondent

Organisation (or individual)	
Single Point of Contact name and contact info	
Number of Technologies (or ensemble of technologies) submitted in this Response	
Number of files included	
List of files included (filename, description)	

For each Technology submitted

Name of the Technology	
Description (High-Level)	
Type (Best of 1 to 4)	
Is available / implementable now? If not, when?	
IPR Declaration	
Licensing Statement	
Use Cases that are addressed by this Technology	

For each Use Case addressed by this Technology

Use Case:	*Use Case* and *UC accronym*	
Requirement	**Fulfilment**	**How / Specifications**
UC-1	(*Fully*, *Partly* or *Not*)	
UC-2		
...		

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Annex C - JT-NM Vision / Mission and Timeline

Project Summary:
(A short summary of the project.)
The Joint Task Force on Networked Media has been created to help manage the transition from broadcast infrastructures that are based on specialty broadcast equipment and interfaces (SDI, AES, etc.) to IT-based packet networks (Ethernet, IP, etc.). This effort spans the entire professional media industry and all of its applications including live and file-based. We intend to accomplish this objective by collecting business-driven user requirements, releasing a Request for Technology, and then by publishing the results of a gap analysis between the user requirements and the results of the RFT.

Sponsors:
(Entities that are responsible for the Task Force.)
The sponsors of the Task Force are the European Broadcasting Union (EBU), the Society of Motion Picture and Television Engineers (SMPTE), and the Video Services Forum (VSF).

Vision:
(A statement based in the future, assuming that the effort is successful.)
New business opportunities are enabled through the exchange of professional media, including file-based and live content, across a network taking advantage of the benefits of IT-based technology at an affordable price.

Mission Statement:
(A statement that describes what the effort will accomplish.)
In an open, participatory environment, help to drive development of a packet-based network infrastructure for the professional media industry by bringing together manufacturers, broadcasters and industry organizations (standards bodies and trade associations) with the objective to create, store, transfer and stream professional media.

Objectives:
(The main thing the effort seeks to achieve.)
The primary objective of this Task Force is to identify gaps that exist between user's business driven requirements for a packet-based network infrastructure for professional media, and the responses from manufacturers when queried about their ability to fulfill the user requirements. Other objectives include promoting interoperability in packet-based systems (networking, equipment and software) for professional media. The ultimate objective for the industry is to help manage the transition between broadcast infrastructures that are based on specialty broadcast equipment and interfaces to an agile, on-demand, packet-based network infrastructure designed to support a variety of distributed, automated, professional media (file- and stream-based) workflows for local, regional and global production supporting any format, standards-based, for interoperability to facilitate new workflows and reduce total cost of ownership and to speed-up content time-to-market.

Method & Approach:

The scope of work of the Task Force is as follows:

- Collect business-driven use cases and requirements to help the industry prioritize and to focus efforts. Publish these use cases
- Issue a Request for Technology (RFT) in order to collect information about technology that can be used to meet the challenges posed by the use cases collected above.

- Look for areas where there are unmet user requirements, and publish these unmet requirements as a gap analysis report, along with the complete text of all RFT responses
- Other work items as defined by the above tasks
- Evaluation point: validate that the Task Force has achieved the items in the scope of work above

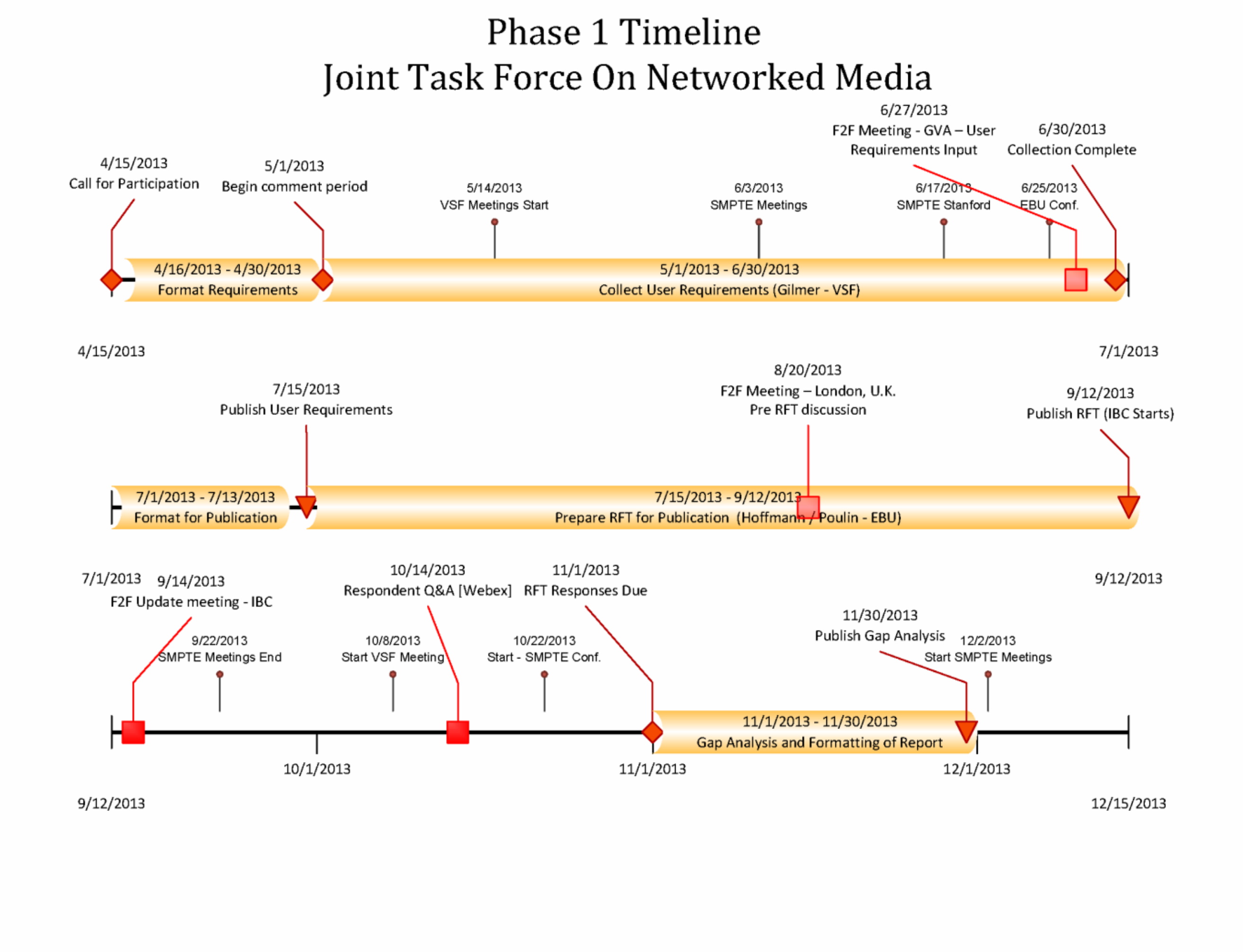
Based upon the successful accomplishment of the scope of work above, the sponsoring organizations will evaluate industry needs and potential future areas of work.

Out of Scope

The following areas are *Out of Scope* for the Task Force:

- The Task Force will not write standards
- The Task Force will not work on signal processing/transformation
- The Task Force will not define Universal Codecs
- The Task Force will not be an *exclusive* group
- The Task Force will not duplicate work done by other groups

The following is an overall project timeline of the JT-NM:



Annex D - List of participants in the *Task Force*

JT-NM Administration Team

Brad Gilmer – VSF
Richard Friedel - FOX - co-chair
Hans Hoffmann – EBU - co-chair
Felix Poulin - EBU
Bob Ruhl - VSF
Peter Symes - SMPTE - co-chair

RFT Management Team

Markus Berg – IRT
Thomas Edwards - FOX
Brad Gilmer - VSF
Al Kovalick - Media Systems Consulting
Sonja Langhans - IRT
Felix Poulin - EBU - Leading
Bob Ruhl - VSF
Karl Schubert - TechNova Consulting LLC

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Annex E - Introduction to the Broadcast Plant - For Networking and other IT Professionals (informative)

This annex is provided solely to aid those who may not be familiar with the professional media industry, and it is meant only as a brief introduction. This is an example and not meant as an exhaustive representation of all use cases. It is for information only and nothing in this annex conveys any requirements upon an RFT Response.

E1. The Typical Broadcast Plant Today

Figure 1 shows a highly simplified schematic of a typically television broadcast plant. The connection lines with arrows represent coaxial cables carrying HD-SDI video or AES audio streams.

The large cross point video router is represented as a “U”-shaped device in this schematic, but physically it is a monolithic box (or aggregation of smaller routers) with hundreds or thousands of coaxial cable BNC connections for audio and video. The router allows video or audio from any input to be routed to any output. Router control panels are found throughout the plant to set up connections between audio and video sources and destinations. Signal path provisioning is done out of band in contrast to IT networks that tend to use in band MAC and IP addressing.

The video server is a device for recording and playing back video and audio streams to/from a storage system (typically spinning hard drives). Servers tend to store video in a compressed format using MPEG-2 or H.264 at 20 – 50 Mbit/s or higher data rates.

A video switcher (also known as a “vision mixer” in Europe) is a device used to select between several different video sources and also to composite/mix them together in effects. Some effects include “fade to black”, “crossfade” between two video streams, “picture-in-picture”, and 3D video warping.

A Closed Caption inserter allows for the insertion of live or recorded Closed Caption data into the video stream. In different countries, subtitle or teletext data may also be inserted.

A logo inserter is a simple video compositor that adds graphics onto a single video stream.

An audio mixer performs the mixing together, panning and filtering of multiple audio sources, generally resulting in a 5.1 channel or stereo final audio mix. Video switchers may also perform some audio mixing during their video effects (such as crossfade).

A distribution encoder compresses the video into a low-bit rate for distribution to cable head ends or broadcast television station affiliates, directly to viewers over the air, or to web users over the Internet. The codec used for distribution is typically MPEG-2 or H.264, and is not designed for a large number of concatenated decode/recode cycles. MPEG-2 distribution bit rates of 15 - 30 Mbit/s are typical for HD video.

Not shown on the diagram is the automation system, which has several time-driven lists of events that control a large number of devices in the plant, especially the video server recording and playback and the video switcher. Automation lists can also be stepped forward under manual control, for example in response to a manual button press that indicates the break in action of a sports event in order to run an advertising spot. Automation has previously used RS-422 serial connections to control devices, but most manufacturers are now moving to IP control.

Monitoring includes simple displays of a single video channel, devices that automatically sense video presence and audio quiet to warn of errors, devices that monitor audio loudness, and “multiviewers” that combine and scale several different video channels on to the display of a single (though often large) video monitor.

Also not shown on the diagram is an audio intercom system which allows broadcast personnel to communicate with one another even if they are located in different locations (for example, between a production control room and the inside of a studio). A special intercom circuit that goes into an earpiece worn by talent is known as IFB (abbreviation for interruptible foldback) or talent cueing system for direct cueing of talent.

Audio Broadcasting: A typical audio broadcast plant, like used in a radio station, does look quite similar. There will be a live/recorded signal coming in, by satellite, ISDN (if still available), via Audio over IP (e.g., EBU Tech 3326) or out of a audio console in the studio. Internally, the audio is transmitted as AES3, using crosspoint audio routers.

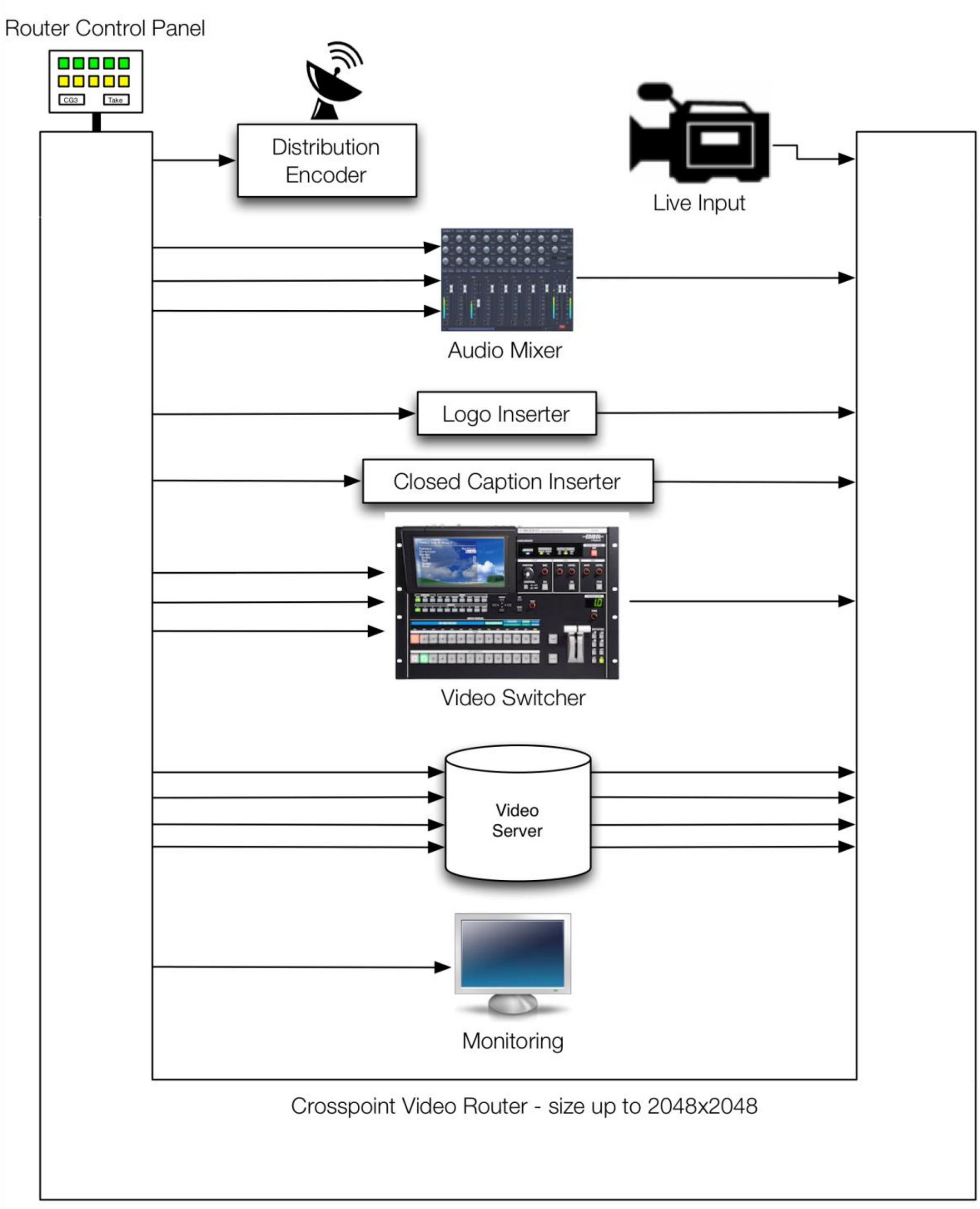

Figure 1: Simplified Schematic of Typical Broadcast Master Control

E2. Today's Digital Audio and Video Standards

High-definition (HD) video tends to be found in two primary formats for television viewing, although frame rates differ by country. “1080i” is an interlaced image of 1920 pixels by 1080 lines. Each frame is formed by painting two sequential fields, one of odd lines and one of even lines. This format is defined by SMPTE 274:2008. “720p” is a progressively scanned image of 1280 pixels by 720 lines, defined by SMPTE 296:2012. In 60 Hz power countries that have used the NTSC standard, 1080i has a frame rate of 29.97 fps, and 720p has a frame rate of 59.94 fps. In 50 Hz power countries, 1080i has a frame rate of 25 fps and 720p has a frame rate of 50 fps.

The typical transport of uncompressed HD video signals in the broadcast plant is standardized as SMPTE 292:2008 (“HD-SDI”), a bit-serial data structure with an on-the-wire rate of about 1.485 Gbit/s. HD-SDI is usually carried over 75Ω coaxial cable, although there is an optical fiber version (SMPTE 297:2006) used mainly in very long-distance connections. Uncompressed 4:2:2 chroma subsampled HDTV signals are transmitted on HD-SDI with 10-bit resolution of each component. Extra “ancillary” data space in the HD-SDI signal outside the active video area can carry Ancillary data packets (“ANC packets”), which can include embedded digital audio signals and Closed Captioning data. The HD-SDI channel coding scheme is scrambled NRZI, and there is a CRC calculated for every active video line. The SMPTE 424/425 standards describe other versions of SDI including a 2.970 Gbps rate for carrying 1080p60 signals. This signal type is often referred to as “3G SDI”.

Digital standard-definition (SD) video is also found in two primary formats, though generally based on country of use. In NTSC countries, the standard is “480i”, a 720 pixel by 480 line interlaced image at 59.94 fps. In non-NTSC countries (where PAL and SECAM systems exists), the standard is “576i”, a 720 pixel by 576 line interlaced image, generally at 50 fps. Digital SD video formats are defined by ITU-R BT.601-4. These signals also are typically sent as 4:2:2 subsampled 10-bit samples. SD video can be carried digitally on SMPTE 259:2008 (“SDI” or “SD-SDI”), a bit-serial data structure similar to HD-SDI, but at a lower bit rate (generally 270 Mbit/s), again over 75Ω coaxial cable.

SMPTE 299:2009 defines the embedding of AES3:2003 24-bit PCM digital audio channels (typically sampled at 48 kHz) in horizontal ancillary data space (HANC) of HD-SDI. Embedded audio is particularly convenient to avoid audio/video lip sync problems from timing differences between audio and video signal paths. SMPTE 272:2004 defines a similar embedding of AES audio into HANC of SD-SDI.

AES3 audio can also be found in the broadcast plant by itself on audio channel pairs using 75Ω coaxial cable (AES-3id). Broadcast video routers often have dedicated audio inputs and outputs, and some also use AES10 Multichannel Audio Digital Interface (MADI) that can carry up to 64 channels of digital audio on a 100 Mbit/s interface. Routers can have internal audio mux/demux capability to extract or insert audio channels from/into video streams, or external devices can do this.

E2.1 Synchronization and Timing

Traditional broadcast plants tend to operate with many separate video signals whose frames and raster scans are synchronized in time. This greatly aids the process of switching and mixing video signals in real time. Switches can be done during the “vertical blanking” period of video outside the typical visible raster (avoiding visible “glitches”), and pixel values from the same video positions from two or more different streams can be mixed together in real time without buffering.

A master synchronization clock is typically distributed across a broadcast plant. This can be done using an analog composite SD “bi-level sync”, a signal indistinguishable from an all-black television signal of the same format, commonly known as “reference black” or “black burst”. A “tri-level sync” based on analog HD signals of SMPTE 2742:2008 or SMPTE 296:2012 is also sometimes used. These analog synchronization signals tend to be distributed on coaxial cable through low-latency distribution amplifiers. Regardless of the means to sync, every AV device in a building/campus can be sync aware and can send or receive video/audio signals synchronously aligned to other facility signals.

Some video processing devices (such as complex mix effects) introduce significant delay to video signals. Video signals can be buffered by “frame synchronizers” that are tied to the master sync signal to briefly delay the signal and re-synchronize it to the signals in the rest of the plant. Some video routers also have frame synchronization capability.

Recommended switching points in the video raster are described by SMPTE RP 168:2009. A standard coding of video frame times is described by SMPTE 12-1. Audio/video synchronization and timing signals based on the IEEE 1588 Precision Time Protocol is now part of standards or draft standards from SMPTE, AES, and IEEE.

E2.2 Uncompressed Video Networking Standards

SMPTE 2022-6 defines the transport of HD-SDI over IP using RTP/UDP/IP (RFC 3550). SMPTE 2022-5 defines an optional Forward Error Correction (FEC) stream that can provide varying levels of protection against RTP Datagram packet loss for SMPTE 2022-6 streams.

IETF RFC 3497 also describes a carriage of HD-SDI video using RTP, and the Pro-MPEG Forum Code of Practice #4 (CoP4) further expands on RFC 3497. Unlike SMPTE 2022-6, all data in each RTP datagram must come from only one line of video, and there is more metadata in CoP4 than SMPTE 2022-6 (such as line numbering).

A draft amendment to IEEE 1722 defines the carriage of HD-SDI and SD-SDI payload formats over Ethernet AVB transport protocol (Audio Video Transport Protocol, AVTP). The mapping is known as the AVTP Professional Video Format (APVF).

E2.3 Uncompressed Audio Networking Standards

Draft AES67 defines the transport of PCM coded audio with sampling frequencies of 44.1 kHz and higher and resolution of 16 bits and higher.

Draft IEEE 1722 defines the transport of IEC 61883-6 digital audio as well as more generalized PCM coded audio over AVTP for Ethernet AVB.

E2.4 Compressed Video over IP Standards

The use of a “visually transparent” codec for video provides significant bandwidth savings over uncompressed video. For example, HD video for contribution over WANs is often encoded using JPEG 2000 at bit rates from 100-150 Mbps, an order of magnitude less bandwidth than uncompressed 1.485 Gbps HD-SDI.

The preferred standards set for JPEG 2000 over IP consist of:

- JPEG 2000 Part 1 Amendment 3 (ISO/IEC 15444-1:2004/Amd3) “Profiles for Broadcast Application” carried in MPEG-2 transport streams (ISO/IEC 13818-1:2007/Amd5)
- “Transport of JPEG 2000 part 1 video over MPEG-2 TS”, carried as RTP/UDP/IP as per SMPTE 2022-2 “Unidirectional Transport of Constant Bit Rate MPEG-2 Transport Streams on IP Networks”
- Optionally using a FEC stream as per SMPTE 2022-1 “Forward Error Correction for Real-Time Video/Audio Transport Over IP Networks”.

The SMPTE 2042 standards also describe “VC-2”, a low-delay wavelet codec intended for professional use.

E3. Conclusion

The existing serial digital architecture for broadcast works fairly well today for rarely changing workflows using SD and HD resolution. The data rates involved and reliability requirements of the architecture are both very high. However it is unclear if the specialized and inflexible hardware of today’s broadcast plant will be able to affordably meet the ever changing workflows and resolutions that the industry expects to tackle in the future. Thus the industry desires networked solutions for professional media.

End of document.