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Final Report on ATSC 3.0
Next Generation Broadcast Television

ATSC Planning Team 2

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The Advanced Television Systems Committee, Inc., is an international, non-profit organization developing voluntary standards for digital television. The ATSC member organizations represent the broadcast, broadcast equipment, motion picture, consumer electronics, computer, cable, satellite, and semiconductor industries.

Specifically, ATSC is working to coordinate television standards among different communications media focusing on digital television, interactive systems, and broadband multimedia communications. ATSC is also developing digital television implementation strategies and presenting educational seminars on the ATSC standards.

ATSC was formed in 1982 by the member organizations of the Joint Committee on InterSociety Coordination (JCIC): the Electronic Industries Association (EIA), the Institute of Electrical and Electronic Engineers (IEEE), the National Association of Broadcasters (NAB), the National Cable Telecommunications Association (NCTA), and the Society of Motion Picture and Television Engineers (SMPTE). Currently, there are approximately 160 members representing the broadcast, broadcast equipment, motion picture, consumer electronics, computer, cable, satellite, and semiconductor industries.

ATSC Digital TV Standards include digital high definition television (HDTV), standard definition television (SDTV), data broadcasting, multichannel surround-sound audio, and satellite direct-to-home broadcasting.

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1 EXECUTIVE SUMMARY

ATSC's Planning Team 2 (PT-2) was charged with exploring options for a next generation broadcast television (NGBT) system – called "ATSC 3.0" – including candidate technologies, potential services and likely timeframes, and without requirement for backward compatibility to current ATSC systems.

PT-2's explorations identified several potential technology components for ATSC 3.0, including improved audio and video codecs, and more robust and/or efficient transmission methods. Most promising among new codecs is current work by MPEG on "High Efficiency Video Coding" (HEVC, also referred to as ITU H.265). Intriguing efforts in modulation and transmit and receive antenna system design have been conducted under the auspices of DVB, NHK and others.

Another important transmission variant to be considered is so-called "hybrid" service, in which over-the-air and online methods of content delivery seamlessly converge at the user's terminal device. While this functionality is already being explored in the context of the ATSC 2.0 initiative (for broadcasting to the most advanced current and near-future televisions equipped with Internet connectivity), it is likely in the ATSC 3.0 timeframe that such capability will have become a standard feature in consumer receivers. Therefore it is considered a fundamental element of the ATSC 3.0 environment.

Several other new or extended applications ("usage models") have also been investigated and found worthy of consideration for NGBT, including content personalization and targeting features, more immersive presentation formats, and advanced non-real-time content downloading services.

Finally, a proposal was made to PT-2 that presented the option for wholesale change in the television delivery mode, in which the use of individual 6 MHz broadcast channels might be replaced by a broadband-style, multiplexed transmission approach, perhaps operated by a third-party service.

In a separate process, PT-2 also engaged with broadcasters, assessing their view of the current state of the broadcasting art, for purposes of assessing their directions and providing guidance in the crafting of subsequent recommendations. Primary sentiments reported here include the need for greater transmission efficiency, increased technological agility, future-proofing of any next generation service, and continued provision of a compelling value proposition in the face of new competitors.

This final report presents some high-level conclusions regarding the desirability of increased quality and quantity of content, increased transmission efficiency, and the value of two-way connectivity in ATSC 3.0 service. The report recommends areas of future study, as well as general timing, business and regulatory factors that inevitably provide a context for any NGBT system.

PT-2 now delivers its findings on relevant technologies investigated, and suggests more detailed examination of some proposed technologies' readiness, along with the probing of certain contextual issues, as a basis for developing a viable next-generation platform for DTV services.

2 SCOPE

The purpose of this report is to present an overview of candidate technologies, as they are currently identified and understood, that might comprise or contribute to a future ATSC broadcast television system. This target system is entitled "Next Generation Broadcast Television" (NGBT), also referred to as "ATSC 3.0." In July 2010, ATSC established its Planning Team 2 (PT-2) to examine NGBT, and ultimately to recommend a potential course of action toward its implementation. PT-2 now delivers its final report herein.

This work is intended to assist ATSC in developing requirements and specifications that move beyond those currently in use, as well as those being considered under the ATSC 2.0 initiative.

The context of this examination has been conducted according to the remit of PT-2 from the ATSC Board of Directors, which presented the group with the following scope of work:

> *"Exploration of potential technologies to be used to define a new/future terrestrial broadcast digital television standard that is not constrained by a requirement to be backwards compatible with ATSC or ATSC 2.0 devices. Analysis (is) to include assessment of the range of services that could be delivered with a new standard and consideration of potential timeframes."*

Therefore, this report considers technologies that are as far-reaching as possible while remaining within the realm of a conceivably viable implementation from today's vantage point.

Notwithstanding the lack of compatibility constraints, however, PT-2's work also has been conducted with at least a nominal regard for a practical migration path for broadcasters, and an underlying intent to retain compatibility where it created no detriment to progress (i.e., avoidance of change for the sake of change).

One area of potential incompatibility outside the control of any standards development organization (SDO) is the allocation of transmission bandwidth, which is necessarily a function of regulatory processes. It is worth mentioning that at least one conceptual system discussed by PT-2 has brought the question forward as to what role might be played by new schemes for allocating bandwidth and/or spectrum aggregation.

So while the technologies presented here may not be backward-compatible to current ATSC systems, they are at least conceptually attainable by broadcasters through some transition from their current operational modes, possibly including the reconsideration of spectrum policy as one variable.

PT-2 made no judgment on whether the ATSC 3.0 system is intended to wholly *replace* or simply *augment* or *extend* existing ATSC technologies, either mobile or fixed. As work progresses within the ATSC 3.0 process, it will be helpful to have a more complete understanding of this issue.

Wherever possible, this report considers new component technologies independently from one other, to allow subsequent work to have the fullest range of flexibility in selecting appropriate candidate components for the ATSC 3.0 system. It also attempts to envision a realistic future environment in which broadcasters will have to continue to strike a proper balance among business, technical and regulatory constraints.

3 INTRODUCTION

To identify candidate technologies in the most exhaustive and enlightened fashion, in lieu of a simple Request for Information (RFI), PT-2 instead conducted a sequence of symposia on the subject of NGBT. Calls for papers to present relevant new technologies were distributed widely to over 80 institutions of learning around the world, and to many vendors and other industry experts. In addition, preceding and during the papers selection processes, invitations were issued to individuals and organizations with knowledge of specific topics of interest and expected relevance to the NGBT. In response to this wide distribution, numerous proposals to present were received.

From these, a volunteer subgroup of PT-2 selected what it considered the most appropriate presentations, and conducted its *First Symposium on NGBT* in Alexandria, VA on 19 October 2010. The same process was repeated leading to the *Second Symposium on NGBT* in Rancho Mirage, CA on 15 February 2011.

In parallel with the processes to present the symposia, PT-2 created a subgroup PT-2A, Ad Hoc Group on Broadcaster Direction, comprising primarily representatives from broadcast stations groups and networks. This AHG was tasked with collecting and summarizing perspectives of existing television broadcasters on the NGBT, including technologies and timeframes. Its report has been issued separately.[1]

Following the two symposia, three more subgroups were formed to examine the specific areas of ATSC 3.0's Physical Layer (PT-2C), Essence Layer (PT-2D), and Integrated Network System Requirements (PT-2E). (PT-2E focused primarily on the "connected" or "hybrid" elements of the ATSC 3.0 environment.)

This report presents synopses of papers and presentations from the two symposia, and the results of subsequent research conducted by PT-2, including resources for further information on the technologies covered. The technologies reviewed are organized under the general subject headings of essence coding, metadata requirements, physical layer, "hybrid" schemes, new usage models, and non-real-time (NRT) delivery.

The report includes a summary of conclusions, recommendations for further study, open questions and other contextual issues that should be considered as the ATSC 3.0 system is developed.

References to all cited presentations appear at the end of the body of the report, followed by Annexes that list the full programs for the two PT-2 symposia, and present tabular detail on various topics considered in the body of the report.

4 TECHNOLOGY REVIEW

4.1 Essence Coding

An ad hoc group on Essence Layer (PT-2D) was formed to analyze essence-related technologies and assess their feasibility. The objective was to research potential future audio and video coding systems; also, essence-related metadata including capability for accessibility-related content (e.g., closed captioning); and to provide a detailed summary and analysis of operational parameters, potential benefits, potential detriments, and other information that may be relevant to an ATSC 3.0 system. The research includes input from the proposals and contributions to the two symposia conducted by PT-2 in 2010 and 2011, and an examination of existing standards as a reference starting point.

The analysis focused on four areas:

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[1] *ATSC PT-2A Broadcaster's Direction Group Ad-Hoc Committee Report*, 14-July 2011.

- Video
- Audio
- Essence-related metadata
- Accessibility-related essence

In each of these areas, the following key attributes were defined (to the extent possible with current knowledge of systems that are proposed or still under development):

- Requirements/key features
- Performance
- Readiness
- Potential applications

Direct recommendations of certain technologies were not provided but instead a general direction of industry was indicated. For video coding, the latest MPEG compression from H.265 (HEVC) shows performance that is substantially better than H.264//MPEG-4 AVC coding. This new MPEG processing continues with the same major blocks, but with a realization of a better use of those blocks. In general, this new coding requires fewer bits to display the same quality level of the images; this trend can be expected to continue.

In audio coding, work is being done for object-oriented coding and efforts toward 3D-audio solutions. Re-creating scenes as realistic as possible with the least amount of needed bandwidth (bits needed) is a common goal.

To get that realistic re-creation across different devices (mobile and fixed) requires local system environment information combined with source metadata. Metadata can be used for a variety of applications including audio loudness control/dynamic range control, closed captioning, sensor data, etc. Given the ever-growing amount and variety of content and display devices, metadata use has increasing allure.

4.1.1 Video Coding

There is general agreement that improvement in video coding can lead to either or both of two outcomes: a codec capable of substantially higher video resolution and/or substantial increase in efficiency. Higher resolution means more pixels per line and more lines per field, as well as possibly more frames per second. Higher efficiency can lead to the same visual quality delivered with less bandwidth, or better visual quality with the same bandwidth, or some combination of both. Both processes can result in better looking images and/or more images within the same delivered bandwidth.

Two Symposium papers [1, 3] were delivered on the High Efficiency Video Codec (HEVC) currently under development in MPEG/ITU, with a target date of January 2013 for the Final Draft International Standard.

The demand for higher efficiency video encoding is driven by a number of different opportunities. Sony [3] suggests that up to ten video sources coded with HEVC could be placed in the same bandwidth as one equivalent source coded with MPEG-2. These multiple video signals could be used in a number of ways. Eyewear-free 3D could benefit where multiple high definition views would be delivered encoded in HEVC. Another potential use is the delivery of several different high definition views of the same program, thus allowing the viewer to decide which aspect or perspective of a program they wish to watch. Still other uses for the freed-up bandwidth were suggested, including multiple HD programs.

The widely varying ideas contained in the presentations with respect to more efficient video coding all lead to the conclusion that a much more efficient video codec needs to be employed by broadcasters in order to free up bandwidth for other uses.

4.1.1.1 Higher Temporal and Spatial Resolutions and Improved Coding

The demand for higher resolution, and thus better visual quality, has historically been a constant. (It could be asked, however, will this demand cease when we exceed the point of human visual acuity?) NHK has proposed its Super Hi-Vision system [4], which is "8K" (7680 horizontal pixels by 4320 lines). Others [1, 2] have proposed a "4K" system (3840 x 2160). The demand for better quality also extends to higher frame rates, higher chroma resolutions in transmission (4:2:2 and 4:4:4 systems versus 4:2:0), increased temporal resolutions (higher frame rates, non-interlaced images), and greater bit depths (10 and 12 bits versus 8 bits). Each of these improvements may impact data bandwidth requirements.

4.1.1.2 Higher Efficiencies

A general conclusion of the two HEVC papers [1, 3] was that MPEG-4/H.264 provides an improvement of about twice the efficiency of MPEG-2. Computer simulations of the currently proposed implementations of HEVC show that HEVC should deliver about twice the efficiency of MPEG-4, or four times the efficiency of MPEG-2. Such increased efficiency can result in reduced bandwidth required for the same delivered quality, or a much higher delivered quality in the same bandwidth.

The need for higher efficiency is not only driven by the demand for higher resolutions and more services, but also for more available bandwidth for enhancements and other services. Beyond the qualitative enhancements discussed above, quantitative proposals include non-real-time file delivery, mobile services, and program-associated data services.

A progression in the development of video coding systems was presented in the First Symposium, and is shown in Figure 1 below:

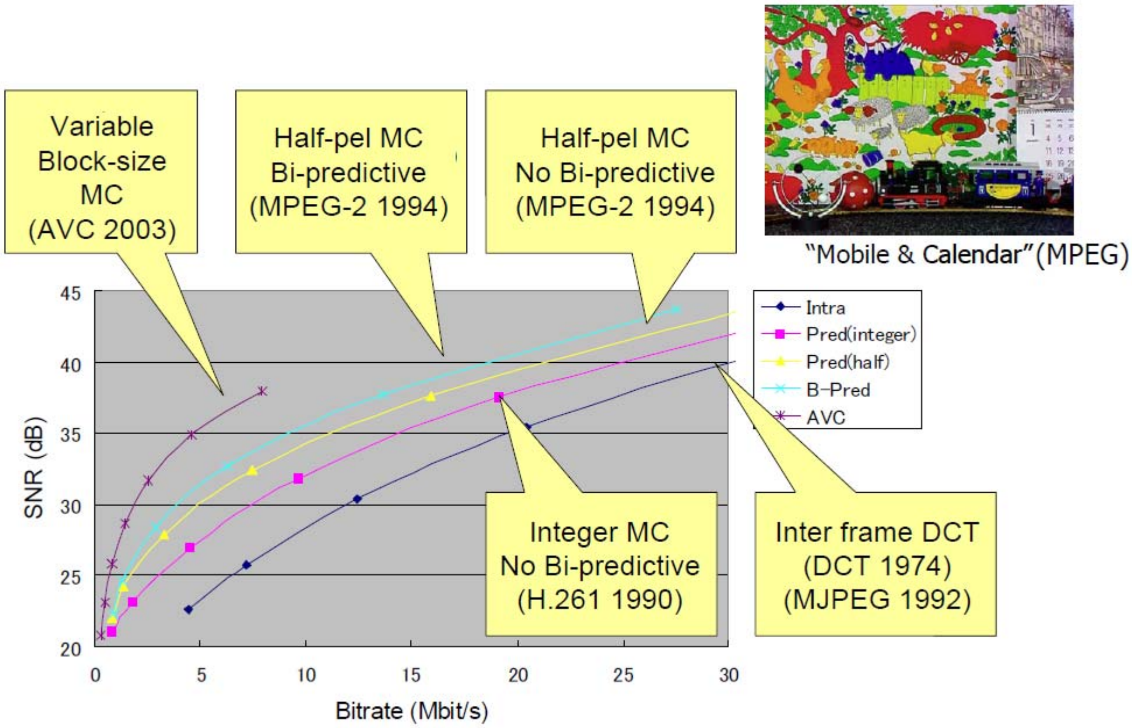

Figure 1: Development history of video coding [1]

Important questions to ask when considering a next generation broadcast television system include the following issues. Whether four times the efficiency of MPEG-2 is enough; whether there will be a successor to HEVC, and if so, what its efficiency will likely be; and, in what timeframe can this hypothetical codec be realized. MPEG-2 was developed in the early/mid 1990s, MPEG-4 AVC/H.264 was developed in the early/mid 2000s, and HEVC is being developed in the early/mid 2010s. Should we expect a ten year cycle before the next generation codec? How does this timetable fit with the development of other aspects of ATSC 3.0, including its practical implementation timeframe? Can we develop an approach that allows for graceful upgrades as they are realized?

4.1.1.3 Video Coding for 3D

Proposals were presented in the two symposia describing additional purposes for new video coding [1, 2, 3]. As noted above these included proposals to facilitate services for glasses-free stereoscopic 3D where, multiple views are delivered. Also proposed was the addition of Triggered Declarative Objects (TDOs) (see Section 4.4.4). It was suggested that a service suite could provide both 2D and 3D services in the same channel. The two presentations on HEVC [1, 3] cited auto-stereoscopic N-view displays as one possible need for and use of this new codec. Multiple views could be delivered in a single channel, or separate depth information could be delivered to allow displays to create the stereoscopic images. Other 3D-related metadata that could be delivered with the additional freed-up bandwidth could include occlusion maps, transparency data and reflectance data.

4.1.2 Audio Coding

Audio coding approaches proposed for expanding the consumer's audio experience have ranged from extending current multichannel systems to higher orders, to object-based coding. Aside from greater immersiveness, the presentation from Fraunhofer [5] also pointed out the increased flexibility achieved with object-based coding approaches such as MPEG-D Spatial Audio Object Coding (SAOC) that would benefit the hearing and vision impaired as well as multilingual support. In particular, it allows the possibility of the viewer having direct control over the dialog level, independently of other program elements.

At the same time, there are practical challenges associated with these new approaches ranging from uncontrolled playback environments and installation complexity, to changes that impact capture and mixing. These approaches and their associated challenges require consideration for next generation broadcast systems.

4.1.2.1 High-order Multichannel Systems

A natural progression for enhancing immersiveness from current broadcast television audio is to increase the number of audio channels. As pointed out in the Dolby presentation [18], systems that support eight discrete channels are commercially available, and the supply of content for these systems continues to grow. There have been several proposals on how to extend the number of channels beyond today's commercially available products (e.g., NHK's 22.2-channel system).

Also suggested in the DTS presentation [6], better utilization of height channels would provide a major improvement from today's commercial multichannel audio systems. In dealing with the "sweet spot" limitation in discrete multichannel systems, increasing the number of channels allows greater envelopment of the audience.

4.1.2.2 Object-based coding

Object-based audio modeling attempts to eliminate the "sweet spot" limitation by allowing multiple audio sources to perceptually emanate from locations within the 3D listening space. With these audio objects it is possible for a single audio source to be perceived on different sides of each listener in an audience (e.g., behind one listener but in front of another; to the right of some listeners but to the left of others), as shown in Figure 2.

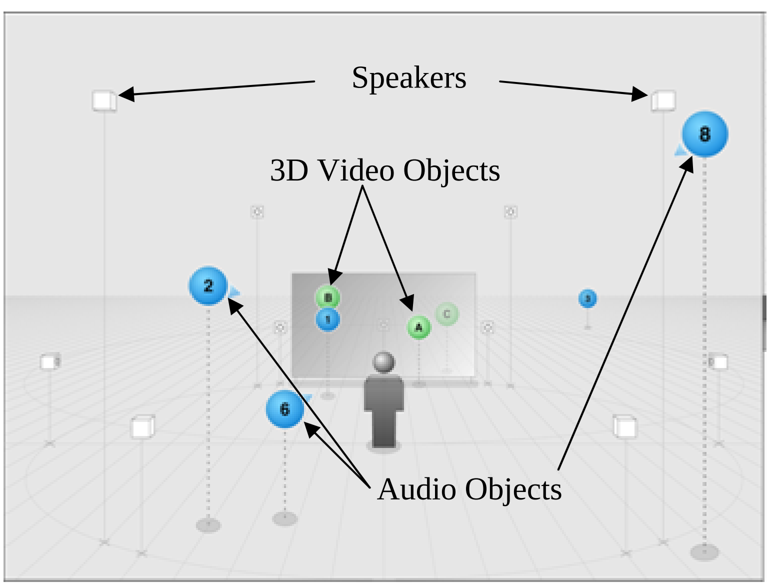

Figure 2: An example of object-based audio modeling [6]

As opposed to discrete multichannel systems, where the angle of each sound source is directly controlled by playing the specific sound out of the speaker(s) most closely positioned to that same angle, these systems use approaches like wavefield synthesis and that described in the DTS presentation [6], where several speakers might contribute to the creation and perceptual positioning of each audio object.

These differences not only represent a departure from classical multichannel audio playback, but potentially from other parts of the ecosystem as well, including creation and mixing. For example, aside from knowledge of audio object locations, there is a greater interest for object-oriented approaches to provide control of more detailed audio effects at the individual stem[2] level rather than at the discrete channel level. Rendering audio objects at playback time provides greater flexibility for interactive applications (e.g., gaming), adaptability and scalability of end-user playback devices, dialog emphasis, etc. These improvements may require new paradigms in content creation, reproduction, and storage and delivery formats.

As noted in the Fraunhofer presentation [5], spatial audio object coding (SAOC) can be implemented without adding untenable complexity to the production process and with only a marginal increase to the transmitted data rate.

In a DTS presentation [8], new general requirements for audio coding are outlined. In order to render audio optimally, it is important to know about the playback environment as well as the content creator’s intent, in real time. There is a need to have enough information/metadata in the bit stream to do this in a smart way (guided post-processing). By adapting to the end point in the home listening environment, information such as speaker configuration, direct vs. dispersive content in each channel, the intended rendering position, and the intended acoustic environment is needed, so that proper rendering can be accomplished.

4.1.3 Essence-related Metadata

Various metadata requirements were expressed or implied within the features presented at the symposia, but there were no presentations on metadata *per se*. This is a critical topic for future discussion and presentations. In modern file-based and workflow-driven systems, metadata is the engine that facilitates all other actions. The fact that no submissions regarding metadata were tendered to the Calls for Papers for either Symposium is a major point of concern, since it could therefore be concluded that it is not top of mind among NGBT developers. PT-2 did not specifically include this topic in its Calls for Papers, and also did not identify the topic during its consideration of invited papers. Going forward, the topic of metadata and workflow as an essential system element should be more specifically included in ATSC 3.0 work, since it may be an important area for extension in NGBT standards.

4.1.4 Accessibility-related Essence

As a new television system is developed it is essential that capabilities be included to accommodate users with sensory disabilities. Captioning and video description were developed for analog television, and then modified for ATSC A/53 digital television, but these components were incremental additions to the developed standards. In the next-generation DTV system

[2] A “submix” group of similar audio tracks, such as dialog, music or sound effects.

design, it is preferable that these capabilities be integrally included from the start, to ensure that undesirable constraints to accessibility are not imposed due to retrofitting limitations or later modifications.

There also may be capabilities beyond captioning and video description to consider within this context, including the possible provision of accessible program-guide information and new emergency alerting features (both notification and content) for disabled users.

The ATSC 3.0 process should place a high priority on all such components as it formulates a next-generation broadcast television system.

4.2 Physical Layer

An ad hoc group on Physical Layer (PT-2C) was formed to analyze various transmission technologies and assess their feasibility. The objective was to research potential future transmission systems and to provide a detailed analysis and comparison to determine potential benefits, potential detriments, and other information that may be relevant to an ATSC 3.0 system. A baseline for this comparison is presented in tabular form in Annex B:.

4.2.1 Revisiting Shannon's Law and Information Theory

In the search for new technology for broadcast transmission, relations to existing technology are required to see the advantages and disadvantages of each new idea. One way to look at comparable technologies with regard to efficient use of spectrum is with a spectral efficiency chart, as in Annex C: below, which shows all major world DTV broadcast and related standards. These include Digital Video Broadcasting - Terrestrial (DVB-T, DVB-T2), Integrated Services Digital Broadcasting - Terrestrial (ISDB-T), Digital Terrestrial Multimedia Broadcast (DTMB), ATSC (fixed and M/H), Long Term Evolution (LTE), China Mobile Multimedia Broadcasting (CMMB), Worldwide Interoperability for Microwave Access (WiMAX), and Forward Link Only (FLO).

Annex C: was obtained from reference values provided in published specifications, other ATSC-related material, and new proposals presented at the NGBT symposia. Such comparison is complicated by the differing transmission methodologies employed across this range of systems, most notably transmission-channel bandwidth variances. Most formats presented to PT-2 in the Symposia assumed retention of a 6 MHz channel bandwidth, and were therefore analyzed by PT-2 on such a common basis. Analysis of other (non-inherently 6 MHz) formats was conducted with normalization to a 6 MHz bandwidth basis.[3]

It is further acknowledged that some of the standards investigated are telecommunications (i.e., 2-way) rather than broadcast systems *per se*, and as a result may be traditionally evaluated with different metrics than those used here. Subsequent evaluation by ATSC may prefer to evaluate systems differently (e.g., comparing systems on the basis of energy-per-bit rather than C/N).

Note that various specifications presented or investigated also used different BER criteria for determining a threshold-of-visibility C/N, as follows:

[3] This should not be interpreted as a recommendation from PT-2 either for or against such channel bandwidth being necessarily maintained in the ATSC 3.0 system.

1. DVB-T: BER = 2E-4 after Viterbi decoder
2. DVB-T2: BER = 1E-7 after LDPC decoder
3. ISDB-T: BER= 2E-4 after inner code correction
4. DTMB: BER = 3E-6
5. ATSC: BER = 3E-6 after RS decoder
6. LTE: BER = “acceptably low”[4]
7. CMMB: BER = 3E-6 after LDPC decoder
8. WiMAX: BER = 1E-6 after FEC decoder
9. FLO: BER = 1E-6 after inner turbo decoder

Any point located near the Shannon limit is regarded as spectrally efficient, and points near the lower left hand corner of the chart in Annex C: are regarded as more robust. For example, a spectrally efficient, robust technology would be DVB-T2 QPSK modulation, but the cost of that robustness is low data rate.

Many of the transmission schemes offer various modes of operation, such as variable FEC, leading to multiple discrete points plotted for a given identified mode. The C/N value is typically derived from performance in an AWGN environment. No attempt has been made to show performance variability across typical RF channel environments (Rician, Rayleigh, Rician-Nakagami, etc.) in the absence of interferers, but it would be helpful providing such values for future benchmarking. New audio and video codecs have become available (MPEG-4 AVC/ITU H.264, for example) that can encode more information into fewer bits. Lower data rates with MPEG-4 AVC/H.264 encoding can have similar quality to higher data rate MPEG-2 encoders. For the physical layer, the capability of higher bit-rates is desirable, but at what tradeoff of reliability? Further comparisons can be seen in [7].

Tradeoffs will always arise when looking for new technologies. When taking steps in evaluating technologies, there should be a pros and cons list for each decision. In the early stages, the beginning steps should be in the direction of the fewest drawbacks. For further reading, refer to the CRC presentation from Yiyan Wu [9].

In general, new technology drives transmission systems closer and closer to the Shannon limit (i.e., toward more spectrally efficient use of bandwidth), but *where* along that Shannon curve is also critical (i.e., high data rate vs. robust channel usage). To satisfy both of these attributes, OFDM-based systems have great appeal, so it is not surprising that many of the new ideas presented to ATSC PT-2 employed OFDM.

4.2.2 New Modulation Schemes

Currently there are two forms of modulation used for broadcasting television signals:

1. Single Suppressed Carrier (e.g., QAM, VSB)
2. Orthogonal Frequency Division Multiplexing (e.g., OFDM, COFDM)

[4] LTE performance values are taken from 3GPP TR 36.942 v9.1.0, in which performance was measured with different bit rates and BER for different modes. [23]

Smarter use of bit mapping to symbols used in these modulations with Bit Interleaved Coded Modulation (BICM), Iterative Decoding (ID) and Signal Space Diversity (SSD) combinations are showing the most efficient spectral use possible to date [9, 10]. Integrating FEC symbols with modulation symbols provides more information that a receiver can use to better demodulate signals, making systems more robust.

Rotating constellations to certain angles and using special bit mappings can also improve performance in bad channel conditions [11]. As shown in Figure 3, by combining all u1 axis data (and separately, u2 axis data) and sending it in different symbols (and on different carriers in OFDM), recovery of more bits can be achieved. For example, recombination at the receiver might correctly recover the u2 axis data, but result in errors on the u1 axis. Recovering at least some of the information helps the FEC improve performance.

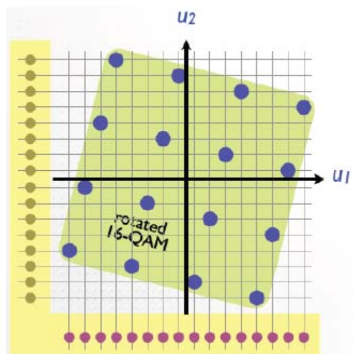

Figure 3: Rotated constellation [11]

4.2.3 New Forward Error Correction Methodologies

DTMB (China) broadcasts and DVB-T2 are now utilizing Low Density Parity Check (LDPC) Forward Error Correction (FEC) coding [7, 11]. LDPC coding enables spectral efficiencies to more closely approach the Shannon Limit [9]. Kyungpook University has studied adding LDPC, together with BCH coding, to existing A/53 8-VSB structures [14]. The results are plotted in Annex C: below, and they show that more spectral efficiency is possible with LDPC and BCH coding added.

FEC coding is the key to delivering high data rates at low C/N values. Further coding techniques were not presented, but new essence coding techniques also reduce data rate requirements, in part via their own improved error correction capabilities.

4.2.4 New Transmission and Reception Antenna Methodologies

Most physical layer presentations proposed use signal diversity (typically, space diversity) as the key for robust reception. Space diversity can have several variations, from many Tx antennas with one Rx antenna, to one Tx Antenna with many Rx antennas. The two approaches currently most favored are Multiple Input/Single Output (MISO, using many Tx antennas and one Rx antenna), and Multiple Input/Multiple Output (MIMO, using many Tx antennas and many Rx antennas) [4, 7, 12, 13, 14].

4.2.4.1 MISO

This technology most usefully applies to mobile devices rather than stationary receivers. It uses one Rx antenna capturing signals from many Tx antennas. This method is being studied at NHK for their Hi-Vision data broadcasting to mobile devices [13]. There are two key technologies needed for MISO operation:

1. Space Frequency Block Coding (SFBC)
2. Good channel estimator

SFBC encoding takes modulated data and passes it to a transmission encoder (SFBC Encoder). Output from that transmission encoder has the original modulated data which is sent to one transmitter antenna and data from the SFBC encoding which is sent to the other transmitter antenna. See Figure 4.

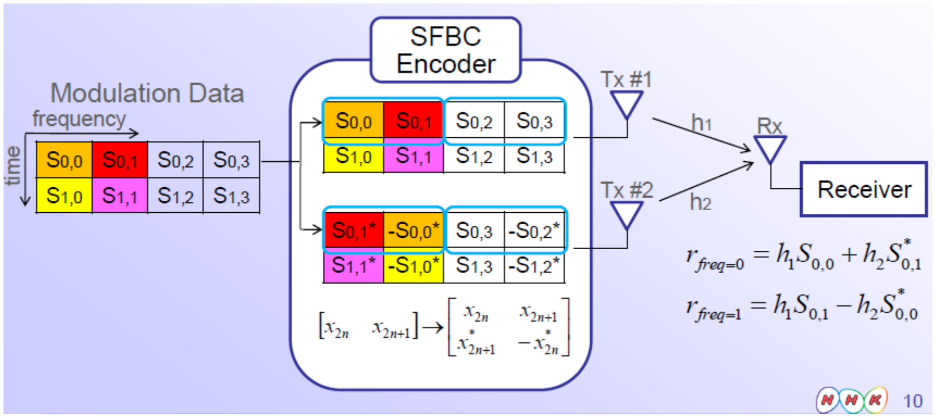

Figure 4: Space Frequency Block Coding [13]

The key aspect in this is to separate out the two data streams in the receiver before SFBC decoding can be done. To do this a good channel estimator is needed. NHK is studying three types of estimators, but all of them use scattered pilots embedded in an OFDM-based signal from each transmission antenna. Those three estimators are as follows:

1. Frequency domain estimator
2. Time domain estimator
3. Frequency/Time combination

To date, for mobile reception, fast estimates are essential, and therefore the Frequency domain estimator appears to provide best performance. Further reading can be done in [13]. This technology is also used in DVB-T2 for Single Frequency Networks (SFN) [11, 12]. Considerable robustness was seen for MISO by Technische Universität Braunschweig in its study of DVB MIMO applications [12].

4.2.4.2 MIMO

Multiple Input Multiple Output (MIMO) systems can be implemented in a variety of ways. A 2x2 MIMO system using Spatial Multiplexing can theoretically double the payload capacity compared to single-antenna transmitters and receivers. But this requires uncorrelated paths between the antennas, which is hard to achieve in practical implementations. Different methods to accomplish this have been tried, such as NHK's cross-polar antenna used for Dual Polar MIMO [4, 13].

NHK requires higher bit-rates for their super Hi-Vision system with 4320 x 7680 video resolution and 22.2 channel audio. NHK's system uses 1024QAM modulation inside the OFDM carriers and dual-polarized MIMO schemes to get 30.2 Mbps capacity per horizontal/vertical polarization. Currently the system works only for line-of-sight stationary receivers. An ISDB-T framework is still being utilized, and C/N reduction is being pursued with better FEC techniques [4, 13].

MIMO attempts to provide higher data rates for transmission with high spectral efficiency, while also providing robustness for mobile receivers. Apparently this only works in theory, however, as indicated by DVB MIMO testing in 2006/2007 [12]. There it was seen that MIMO works well for line-of-sight conditions but has no benefit for mobile reception. For stationary receivers, MIMO works well to increase payloads, but shows only limited gain for mobile receivers due to low SNR. On the other hand, MISO techniques can increase the signal robustness for mobile receivers, as DVB-T2 has shown in SFN networks [11, 12].

Kyungpook University has studied MIMO technology (along with a FEC of LDPC and BCH) and applied it to VSB [14]. It has developed a hybrid version of MIMO using spatial diversity and spatial multiplexing on separate transmitter antennas. The spatial diversity is Space Time Block Coding (STBC), and the spatial multiplexing is Linear Dispersion Coding (LDC), as shown in Figure 5.

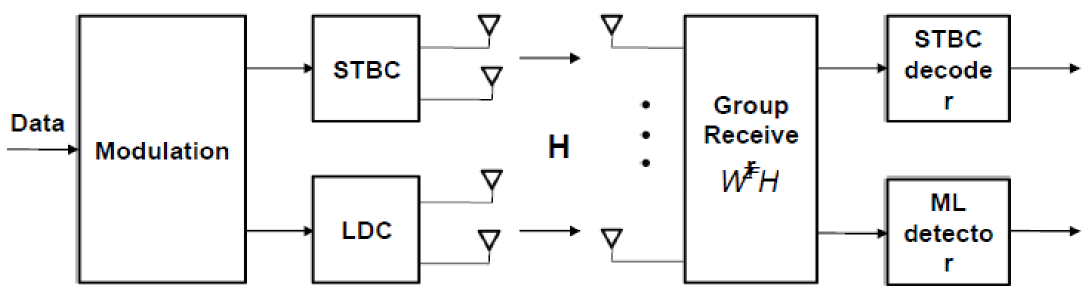

Figure 5: Hybrid Space Time Block Coding system (STBC) with Linear Dispersion Coding (LDC) [14]

Results of this system show a 30.6 Mbps data rate is possible at about 18 dB SNR. The main point from the Kyungpook University presentation is that adding LDPC to 8-VSB as per A/53 is not enough. MIMO techniques need to be added to get close to the Shannon limit and compare favorably with DVB-T2 [14].

4.2.5 Software-defined Radio Applications

Software-defined radio (SDR) is often, and somewhat optimistically, defined as a system in which most of the receiver's RF signal processing is performed in software, using digital signal processing methods, rather than with traditional discrete hardware components such as resistors, capacitors, diodes, etc. Some of the advantages claimed for the SDR approach include flexibility of demodulation modes (new modes can be added easily by simply upgrading software); improved performance over a conventional receiver since digital techniques make it possible to implement sharper selectivity filters and more accurate (i.e., mathematically precise) demodulators and decoders; and, improved consistency and stability because component tolerances and aging do not play as important a role compared to conventional receivers.

The claims for SDR-based RF signal processing for digital television receivers must be examined critically to determine what functions are feasible based on expected increases in semiconductor complexities (Moore's Law) and what are more fundamental limitations. The performance improvements in SDR are obtained in the elements of signal recovery and processing that are performed in the digital domain – i.e., after the A/D converter. Some of the most critical signal recovery circuits, however, are in the "front-end" of the receiver and precede A/D conversion. It is this necessarily analog "front-end" that conditions the signal to meet the limitations of A/D converter technology. For over-air broadcast, the dynamic range of the desired signal is expected, per A/74, to be nearly 80 dB. For new modulation formats, the dynamic range requirements could be even greater, particularly in lower C/N operating modes.

In addition, multiple undesired and potentially interfering signals of amplitudes comparable to or larger than the desired signal can be expected to be present simultaneously, further increasing the range of signal levels that must be processed by the front-end. All of these signals are within VHF and UHF television frequencies. An A/D converter that can directly sample these frequencies with a sufficient number of bits of resolution (16 bits, likely more) is not today a practical device. Said in a different but equivalent way, if an A/D converter is to function directly as a front-end, it must be able to reproduce faithfully all of this desired-signal-plus-interference and devote enough bits to the desired signal that subsequent arithmetic processing (e.g., digital filtering) will preserve adequate S/N within the desired signal.

To put this discussion in the context of current DTV receiver design, it is noted that lower frequency A/D converters can be used successfully as the input to digital demodulators, but they must be preceded by high-quality analog frequency selection and conversion (i.e., an analog front-end). It is currently not possible to obtain the high performance provided by an analog front end in a purely digital SDR.

A few examples of signal conditioning normally provided by the analog front-end:

- "Mixing" the RF signal down to some lower intermediate frequency (IF) or to baseband so that it is at a more practical sampling frequency for the A/D.
- Applying some degree of automatic gain control to reduce the dynamic range (bit width) required for the A/D.
- Applying analog tracking filters around the desired signal frequency to reduce interference and thus reduce the dynamic range required of the A/D.
- Impedance matching to the antenna.

It is expected, therefore, that practically realizable SDRs for the foreseeable future will continue to require analog front-ends of the type and general performance attributes that are found today. Therefore, many of the spectrum planning and interference control issues that are found today will remain. SDR use is likely to grow in signal processing areas where SDRs are technically feasible.

Examples include flexibility in modulation types, in shaping and sizing of the desired signal passband, and some increased degree of removing in-band interference, although it should be noted that the adaptive equalizers and other digital filtering in today's digital receivers already offer some of this capability. The role of software-defined processing in receivers will continue to expand, but work toward this goal must not deny the realities of present technology.

Although SDR equipment is in commercial use today, its cost and applications have limited it to strictly professional service. While the possibility should continue to be examined that SDR technology may become sufficiently cost effective for use in consumer receivers within the timeframe envisioned in this report, sufficient commercial need and application would have to exist to drive such development. For broadcast application, it remains difficult to apply to the high-power transmission environment used in broadcasting (although a distributed “cellular” transmission architecture could conceivably lend itself to SDR application – see Section 4.4.1 below.) At present, sufficient need to generate adequate SDR development does not appear likely for the consumer broadcast receiver environment.

On the other hand, another more commercially viable application of SDR appears in *two-way* communications, where the adaptive channel modeling techniques made possible by SDR are used for configuration of both transmission and reception. This application enables efficient shared access to a given channel, via SDR devices’ awareness of existing users in a given band, and the resulting avoidance of interference with such incumbent users when setting up any new communication channel. (The FCC has studied the possible application of this approach in its *Cognitive Radio* proceedings, and current regulation applies this concept to *DTV White Space* devices, as one example.)

For all of the above reasons, it seems likely that SDR will remain unsuitable for general broadcast use in consumer devices, so it remains an area requiring further study at an appropriate point in the future. This appears to be confirmed by the lack of any specific SDR applications in any of the presentations made to date in PT-2’s exploration of NGBT, despite the inclusion of the SDR topic in the two Symposium Calls for Papers.

The foregoing general discussion of the state of development of SDR notwithstanding, one approach that applies some of the adaptive reconfiguration concepts of SDR to the broadcast model appears in Section 4.2.6. Another tangential application of software-defined receivers is discussed in Section 4.4.5.

4.2.6 Adaptive Transmission Systems

The Electronics and Telecommunications Research Institute (ETRI) is looking into a Self-Organizing Broadcast Network (SOBN) [17]. This idea assumes a feedback channel from receivers and re-configurable network elements, i.e., transmitters that can alter their output power or other parameters (e.g., antenna pattern) automatically upon algorithmically generated commands. Application of this idea is only feasible for SFNs or broadcast networks that have many transmitters to cover an area.

With this model, receivers could measure certain channel characteristics and feedback data to a SOBN server which could control local transmitters to mitigate channel echoes, equalize power between transmitters, and ensure reliable service. Currently this idea has only progressed to the conceptual level.

4.3 “Hybrid” Schemes (Broadcast + Network Connections)

A combination of OTA and other content relay mechanisms is anticipated to be needed in a future broadcast system, including platforms that supplement OTA service with bidirectional interactivity. An ad hoc group on Integrated Network Systems Requirements (PT-2E) was formed to analyze potential future requirements of "connected" television service, including hybrid on-air/online content delivery, user interactivity, adaptive configuration, and other enhancements enabled by broadcast television operating in an integrated, networked environment. One of the group's objectives was to research potential future hybrid systems and to provide a detailed analysis and comparison to determine feasibility, potential benefits, potential detriments, and other information that may be relevant to an ATSC 3.0 system.

It also should be acknowledged that concept of hybrid broadcasting has been proposed for future inclusion in the ATSC 2.0 process, as well. For ATSC 3.0, therefore, appropriate extension of possible existing capabilities in ATSC 2.0 will need to be taken into account.

There have been various schemes under development to support delivery of broadcaster content using a coordinated combination of in-band (i.e., through the OTA RF transmission) and out-of-band (e.g., Internet, etc.) channels.[5]

These hybrid broadcast schemes include, but are not limited to:

- HbbTV
- Hybrid-Cast
- Media Fusion
- MPEG Media Transport (MMT)
- OHTV

Looking at today's TV user experience, there are four main points of interest:

1. Digitization of TV: The public is accustomed to high quality and reliable TV picture for a low or zero cost.
2. Growth of the Internet: Social networking and user-generated content is appealing.
3. Increased use of broadband: Video-on-demand options, and personal customization with high-speed and reliable Internet access has become the norm.
4. Diversification of connected devices: Smart phones can now offer web-browsing and video playback while user is mobile. The public has become accustomed to watching content while mobile, all on a variety of products.

4.3.1 HbbTV

Hybrid Broadband Broadcast TV (HbbTV) is a pan-European specification, based on HTML and web technologies, targeted to hybrid terminals (e.g., connected TVs) that receive an over-the-air transmission and can be connected to the Internet via a broadband interface.[6] Founding Members of the initiative include ANT, Astra, France Television, IRT (representing ARD and ZDF), OpenTV, Philips, Sony, Samsung, TF1, and EBU. The HbbTV specification unifies a number of existing technologies and specifications:

- Open IPTV Forum (OIPF)
  - JavaScript APIs for TV environment (e.g., tuning, PVR, etc.)
  - Media formats (& protocols)

[5] The material on HbbTV, MMT and OHTV described herein was supplied by ATSC PT-3 [19, 20, 21].

[6] Version 1.1.1 of the HbbTV specification was approved as *ETSI TS 102 796* in June 2010.

- CEA
  - JavaScript APIs for video streaming
  - Subset of W3C specifications & image formats
  - Remote control support
- DVB
  - Application signaling
  - Application transport (DSM-CC)
  - Stream events
  - DVB URI
- W3C
  - XHTML
  - CSS 2.1, CSS-TV
  - DOM-2
  - ECMAScript
  - XML HTTP Request

The HbbTV specification was approved by ETSI as a standard in June 2010. [22] More than 60 companies support the standard, including UK DTG. Products and services have been deployed in Germany, and a rollout has been planned in France.

A presentation supplied to ATSC PT-3 by Nagra and OpenTV suggests the possibility of a formal liaison between the HbbTV consortium and ATSC, for exploring potential harmonization of future technical specifications [19].

4.3.2 Hybrid-Cast

NHK's Hybrid-Cast is a system that combines Internet features and broadcast content [15]. To make the system work, there needs to be close synchronization between broadcast and broadband content. The concept of Hybrid-Cast is illustrated in Figure 6.

Interaction with the program content is the motivation. For instance, while watching a movie the user can text a comment about a certain actor in the film. Hybrid-Cast receivers can then pull up all movies in which the actor appears, and list them in a recommendation area for user browsing. Other envisioned services include program customization, social TV, program recommendation, and multi-device linking. System specifications are now being defined, and a prototype set-top box receiver is under development at NHK.

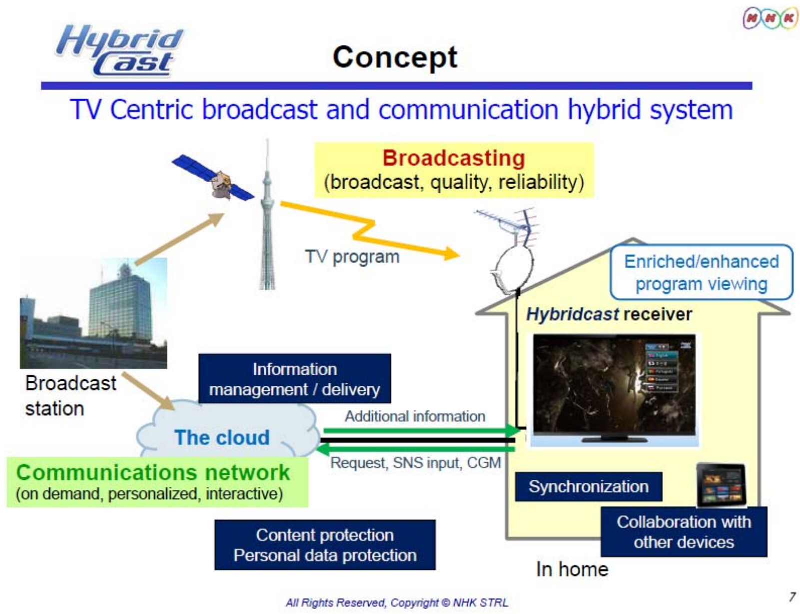

Figure 6: NHK Hybrid-Cast system [15]

4.3.3 Media Fusion

Sony presented its "Media Fusion" concept that integrates different types of media (broadcast, IP networks, cell-phones, etc.) [2]. Four service use cases were identified. Combining media and enabling different media types to communicate with one other allows for more user features, such as:

1. Fixed/ Mobile Device Interaction
2. Targeted Ad switching
3. Interactive Video Portals to the internet
4. Free-Viewpoint Service (user defined angles of viewing)

Fixed/ Mobile device interaction could allow mobile devices to control stationary TVs and/or TV content, via display of an Electronic Program Guide (EPG) on the mobile device. Targeted Ad switching enables region- or product-specific advertisements, such that these ads are displayed only to most likely interested consumers. Internet access from a TV is enabled through an Interactive Video Portal. Another example of such converged media is user-defined viewing angles being generated with a Free-Viewpoint Service. Some of these services are further described in Section 4.4.

4.3.4 MPEG Media Transport

In 2010, MPEG launched a new standardization work item, called MPEG Media Transport (MMT). The main objectives of MMT is the efficient delivery of media in an adaptive fashion over various networks, with the main emphasis on IP-based networks, including terrestrial, satellite and cable broadcast networks. The standard will enable building interoperable solutions for delivery and consumption of media in this context.

MMT also enables the use of cross-layer designs to improve the Quality of Service/Experience (QoS/QoE). By incorporating QoS/QoE-related information from different layers, the delivery and consumption of media would be optimized.

The specification will provide the capability of seamless and efficient use of heterogeneous network environments, including broadcast, multicast, storage media and mobile networks, and enable bi-directional, low-delay services and applications, such as online gaming and conversational services.

It is also intended to enable efficient signaling, delivery and utilization of multiple content protection and rights management tools, as well as efficient content forwarding and relaying efficient one-to-many delivery, and a means for error immunity, including burst errors.

MMT can be divided into three functional areas, namely: Encapsulation, Delivery and Control. Encapsulation will define the format to encapsulate encoded media data either to be stored on some storage device or to be carried as a payload of delivery protocols. Delivery provides functionalities that are required for transferring encapsulated media data from one network entity to another. Control provides functionalities to control the media delivery and consumption.

A document supplied to ATSC PT-3 remarks that “ATSC needs to follow if this MPEG technology can satisfy what ATSC needs for Internet Enabled TV, and should consider it for one of candidate technologies for Internet Enabled TV if the timeline is aligned.” [20]

4.3.5 OHTV

Open Hybrid TV (OHTV) is a new TV service for TV receivers with a broadband network connection [21]. The OHTV Standard is in progress under the auspices of the Korea Next Generation Broadcast Forum. OHTV considers various service scenarios using both broadcast and broadband connections. See Figure 7.

The Founding Members of OHTV are:

- Broadcasters : KBS, MBC, SBS, EBS (Korean broadcasters)
- Manufacturers : LGE, Samsung, Net&TV
- Academia : Realistic Ubiquitous IPTV ITRC (Kyung Hee University)

Eight main services were determined (from a survey of all participants). The top five of those services have been standardized in a first phase. Prototypes were demonstrated at the NAB 2010 Conference and the KOBA Show in 2010.

- NAB : Push VoD (NRT), IP VoD, Video Bookmark
- KOBA : Push VoD (NRT), IP VoD, Advanced EPG, Video Bookmark

Co-promotion celebrations/demonstrations of broadcasters were held in December, 2010. OHTV specification has been approved as draft document to TTA, and OHTV Implementation Guideline will be developed in 2011.

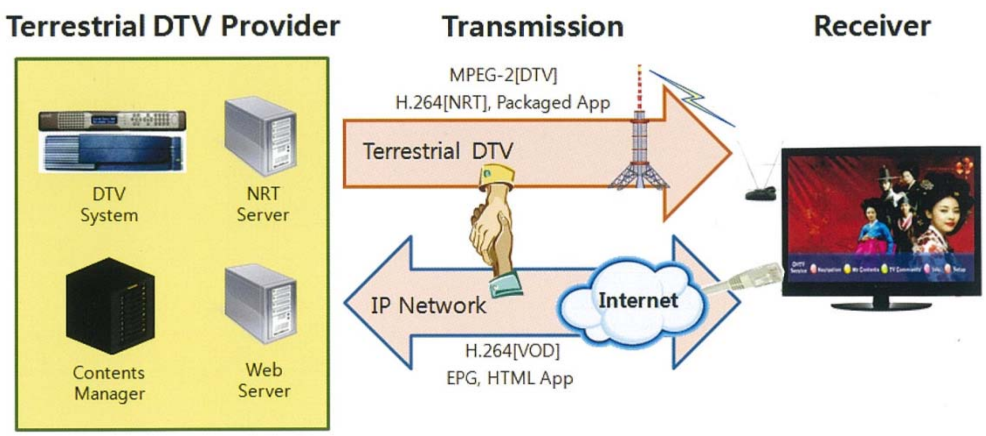

Figure 7: OHTV System [21]

The OHTV technology is based on ATSC and Web Technology:

- Broadcast Technology : ATSC, PSIP, ATSC-NRT
- Web Technology : HTTP 1.1, W3C Widget, HTML, CE-HTML (CEA-2014), OIPF DAE (Open IPTV Forum Declarative Application Environment)

A document submitted to ATSC PT-3 [21] suggests that the OHTV Standard could be incorporated into ATSC 2.0 by using the following:

- ATSC PSIP extension: a "link descriptor" describes what kinds of resource for Advanced EPG should be linked.
- Hybrid content delivery over NRT and Internet: Push-VoD content can be delivered through ATSC-NRT and HTTP to reduce the broadcaster's maintenance cost of CDN.
- A triggering event that plays downloaded content and activates applications on main video (e.g. voting application, T-commerce application).

4.4 New Usage Models

It is critically important that ATSC 3.0 address (and ideally, leverage) the context that broadcasters – and future audiences – will exist within. Among the new elements foreseen in this environment are the following:

4.4.1 New Networking Topologies for Content Distribution/delivery

While not a wholly new concept, a move away from traditional single-channel-per-carrier services toward multiplexed or otherwise aggregated transmission may become more feasible in the future. A presentation from Rohde & Schwarz and Sinclair Broadcast Group [16] proposed one such "universal broadband broadcasting" concept. In this proposal, both taller "single stick" transmitters and cost-effective single-frequency-network (SFN) overlays could be used. This would be a next generation broadcast television system designed with a capability to support the emission of multiple OFDM-symbol frame structures. The transmission mode of data is broadcast or multicast only, and could include priority traffic from Homeland Security and or that of first responders to help serve the public interest. This aggregated data is assigned to the appropriate transmitter/s and could be also used to “off load” common and high demand data (video being a large capacity driver) from MNOs (mobile network operators or wireless carriers).[7]

It was proposed that the allocation of OFDM carriers in the broadband signal could change dynamically to accommodate the opportunity to control the spectral footprint for a variety, of reasons. This could enable the dynamic sharing of spectrum, and provide possibilities to shape spectrum occupation, thereby enabling more efficient use of spectrum, while mitigating interference. Furthermore, the basic allocated blocks of spectrum, possibly up to 20 MHz each (instead of the present 6 MHz), would be channel coded and modulated, then feed to broadband transmitter(s) to achieve increased spectral efficiency and/or other advantages. If desirable, several such blocks could occupy a contiguous portion of the UHF band by aggregation of multiple blocks.

The proposal suggested working towards convergence with IMT-Advanced (LTE, IEEE 802.16M) systems, if possible, by operating collaboratively within portions of the broadcast television spectrum. This is envisioned by leveraging what was described as a “Universal Frame” structure, much like the “Future Extension Frame” concept documented in DVB-T2. Portions of OFDM carriers of a block transmitted from a given site (service edge) could be finely adjusted (for example, to reduce interference) by suppressing particular Orthogonal Frequency-Division Multiple Access (OFDMA) carriers or by enabling their sharing using an envisioned time division multiplexing mechanism. This would require most or all users to be under the control of a licensed spectrum sharing database.

The proposed new network topology is shown Figure 8 below. OFDMA offers many more degrees of freedom, and this will allow flexible use for this broadband network architecture. Note the broadband panel antennas (designed beam tilts) direct RF energy downward into the service area, rather than projecting RF energy towards the radio horizon as is done for a single-tower topology to create a coverage area. This may enable frequency re-use at much smaller geographic distances.

[7] The new network topology can use a “Licensed Spectrum Sharing Database” to manage the OFDM emissions from all sites (number carriers as a function of time and geography, etc.) to minimize interference and/or to enable a dynamic spectrum use mechanism. (The design of or the entity that operates database is TBD; it is only put forward in the spirit of spectral efficiency.)

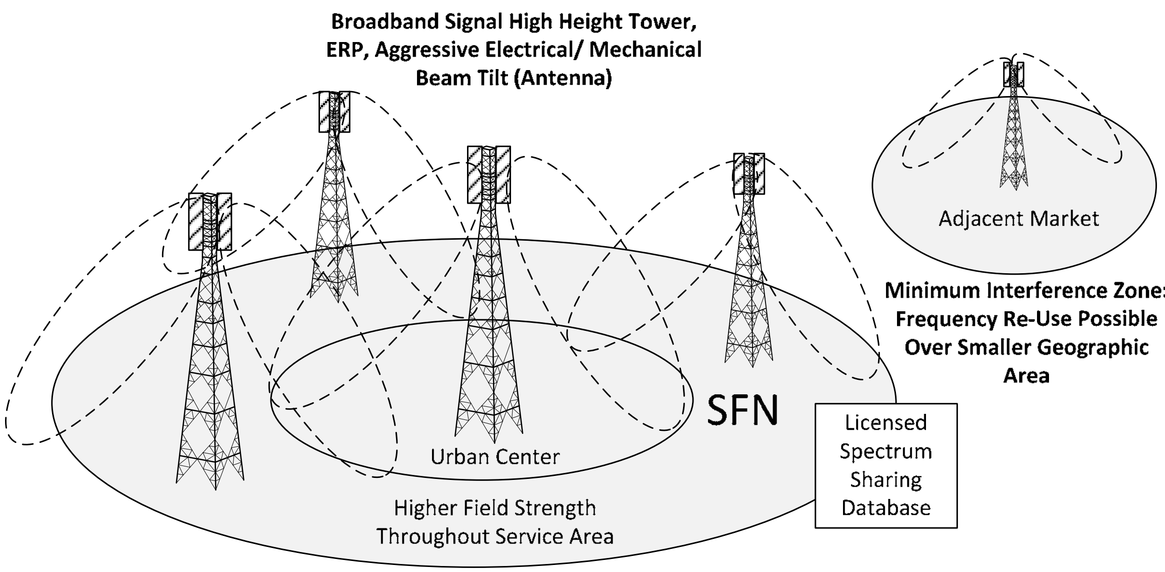

Figure 8: A proposed Broadband/Broadcast delivery model [16]

4.4.2 Personalization

A presentation from Sony [2] offered service use cases in two areas:

1. Effective use of broadband
2. Integration of converged media

Personalized service was introduced under integration of converged media. Examples include viewpoint control (described in Section4.3.3), where increased efficiency of codecs combined with increased efficiency of digital channels makes this possible. This process uses several cameras, each sending information to a receiver that combines the viewpoints and creates the desired angle. Or, control of HD content could be performed with mobile devices, where content information is shared between many devices. In addition, pre-selection of material based on personal preferences, demographics, and interests (PDIs) is possible. Interaction with the Internet provides another service that is easily envisioned today, and NGBT's likely inclusion of an interactive video portal would enable this.

4.4.3 Targeting

The Sony presentation [2] also presented a type of service use case considered under integration of converged media, which provides the option for advertising targeted to specific geographic areas. This type of service can use content from Internet streaming or from stored data content where, in either case, the content selection is based on PDI information. One example is where advertisements are presented to users who are identified as more likely to use the advertised products.

Program-content targeting based on PDIs was also discussed. This selection of content allows for flexible-length programs as users scan through news and information of interest and can reject information they do not want to see. Audio and video signal synchronization needs to be carefully controlled, but delivering only the desired content makes for more efficient use of users time.

4.4.4 Immersive Presentation

A number of proposals for extended or new services that could be described under the umbrella of "immersive" were also discussed during the two symposia. Free-Viewpoint services were suggested where a 4K or 8K video image is transmitted, and the viewer is given the option of selecting an area of that image and displaying an equivalent HD quality image. [2]

In audio, "immersive" presentation extends the traditional surround sound field into 3-dimensional audio space, with height channels and higher-resolution multichannel sound, as well as possibly including flexible, object-based audio reproduction/rendering. [5, 6]

Immersive experiences were also proposed in conjunction with other media ("hybridcasting" as discussed in Section4.3). Some proposals included the addition of Triggered Declarative Objects (TDOs), where execution of the TDO might take the viewer to another medium, or divert them to a different program segment.

Typical use of TDOs involves content- and service-related information being accessed by a user in real-time. These services are only displayed on an access-controlled basis, and are operationally defined by the broadcaster via declarative-content website hyperlinks. For example, portal service can consist of one HDTV (multi-video) service with a TDO as the user selects which camera angle to view. In 3DTV, a TDO can allow a user to select which viewpoint to see a program from, as in the Free-Viewpoint Service described in [2].

4.4.5 "Agile" Receiver Design

Although not addressed specifically by any Symposium presentations, a topic of considerable discussion within PT-2 has been the search for an easy upgrade path to successive ATSC generations. The largest obstacle would seem to be the requirement for consumer hardware replacement when generations change. In this respect, moving from one generation of digital technology to the next would be no less cumbersome than was the transition from analog to digital TV. The "connected receiver" (either as a standalone set-top box, or an integrated television) with receiving and decoding capabilities implemented in firmware and software provides a potential solution to this conundrum, by offering the possibility of downloadable field upgrades, similar to the PC environment. Upgrades could be managed and delivered via the receiver's network connection and, once installed, the device could receive content transmitted in the next-generation broadcast format without the need for consumer hardware replacement.

The significant appeal of such a scenario warrants serious consideration in the current NGBT process, given the flexibility it would provide going forward. Nevertheless, there are substantial obstacles to this approach. The strong requirement for optimum cost-effectiveness in the CE environment could cause reluctance among manufacturers to provide a receiver with the resource "headroom" (additional and initially underutilized processing capability, memory, etc.) that might be required to handle future upgrades. Further, it is always difficult to project the actual needs of future platforms. (Witness the PC world, where software requirements continually outpace hardware capability.)

The advantages of such receiver agility are profound, however, and may be worthy of future exploration. It is likely that the most practical course toward such a goal is through a universally accepted standardization process, which would set the parameters and minimum requirements for such an agile platform. Thus it may be an important component of ATSC's NGBT effort.

4.5 Next-generation NRT

Readers may be aware that ATSC has published a Candidate Standard for NRT (Non-Real Time), referred to as NRT 1.0. NRT is likely to be attractive to consumers in the near future because it provides users with a convenient method to access additional data, receive news updates similar to an RSS feed, and access special content. To review, the capabilities of NRT 1.0 include the following:

1. Browse and Download: A list of content that can be selected for later download
2. Push: Request-based content where a receiver caches content and updates files as needed
3. Portal: Like a web-browser, whereby supporting files of text/graphics are rendered in near real-time

The next ATSC activity in this area is NRT for ATSC 2.0 ("NRT 2.0"). This service integrates Triggered Declarative Objects (TDOs), by which the service provider defines the downloadable content. When the service provider has control over content, the following features can be enabled:

1. Catch-up TV service (allows viewing of missed programs)
2. Advanced program guide, with additional detail on content
3. Service usage reporting

ATSC 2.0 considers the Internet as another data portal for televisions. With this extra data input, the NRT services expand for more user interaction. Work on this is currently in progress in ATSC Specialist Group TSG/S13.

PT-2 believes it is logical to assume that ATSC 3.0 might further extend such features ("NRT 3.0"), but given NRT's as yet untried nature in current generation systems, it is difficult to forecast at this time whether such extensions will be warranted, or what specific form they might take. Therefore PT-2 suggests that decisions on whether and what form NRT might take in the ATSC 3.0 environment be based upon careful observation of NRT's commercial progress in the interim.

5 SUMMARY & CONCLUSIONS

5.1 Potential Timeframes for ATSC 3.0

Perhaps even more challenging than forecasting *what* technologies will arise in the future is specifying *when* they will arrive at sufficient maturity to be deployed for viable consumer service.

PT-2 concludes that technical, business and regulatory developments may all influence the specific timeframes for the ATSC 3.0 era, and therefore recommends that those involved in the ATSC 3.0 development pay close attention to all such developmental contexts as their work proceeds. Additional thinking along these lines is presented in Section 5.3 below.

5.2 Discussion Points for Further Study

PT-2 suggests that the following issues are germane to migration to NGBT, and worthy of early consideration within the ATSC 3.0 development process:

- Will ATSC NGBT services wholly replace or simply augment legacy ATSC services? If the former, which legacy ATSC services (e.g., fixed service, multicasting, NRT, M/H, etc.) are appropriate to augment incrementally in place in the meantime, while technology for total replacement of current ATSC services is developed? (In other words, how exactly does ATSC 3.0 intersect with ATSC 2.0?)

- ATSC 3.0 must have a criterion of providing substantially and demonstrably better service than ATSC 2.0; "substantially and demonstrably better" needs to be defined.
- It is assumed that ATSC 3.0 will incorporate relatively intact any elements of the ATSC 2.0 initiative that have subsequently become successful (i.e., those features that have found substantial broadcaster implementation and consumer usage in the interim). Therefore an assessment of uptake for ATSC 2.0 services will be essential at some point in the ATSC 3.0 process.
- ATSC 3.0 Scope should include not just broadcasters' own OTA application but also the envisioning of the complete content-delivery ecosystem, including MVPDs, Internet, and Wireless Broadband distribution.
- One new area to explore regarding OTA robustness is better understanding of the distinctions between a broadcaster's "coverage" vs. "service" areas. Further study of time and location variability as regards "service area" is a key element of this understanding. Another part of this exploration should involve the concept of scalable forward error connection (FEC) – perhaps applied selectively to different components of service – as well as the effects of various network topologies.
- Scenario planning may be one means of analysis, including at least the conceptual exploration of the following "out of the box" approaches:[8]
    - One scenario should focus on an "all mobile" OTA broadcast (i.e., OTA transmission is targeted exclusively at mobile delivery)
    - Another scenario should consider an "all connected" environment (i.e., every DTV receiver is also an Internet-connected device)
    - A "long tail" business model should also be explored, in which main OTA services carry mass-appeal content, while NRT, online and/or other alternate delivery service from broadcasters carries narrow-interest content.
    - Continuance of OTA-determined territorial boundaries may be required for broadcasters' online distribution under some scenarios (e.g., network-content exclusivity). This may be essential to the use of some hybrid-casting models.
- Envision receivers having location-reporting capability (with user opt-in control likely), and consider leveraging this functionality for sub-localized service features.
- Consider next-generation "Second Screen" (or "Social TV") features.
- The goal of content portability will require solutions for the consumption of that content on a variety of devices, known and unknown Solutions may be needed to coordinate the seamless transfer of content across these devices. Content produced for one medium or transmission path may also need to be managed efficiently over other media and paths.

[8] Note that some of these scenarios may coexist with one another, while others may be mutually exclusive.

This requires an understanding of how content (and metadata) should be packaged to present the same experience to all users on all devices.

- The need for digital rights management, content protection and conditional access will raise considerations involving integrated networks. Solutions may be needed to support content protection in a way that is agnostic to the actual low-level encryption protocol.
- In the context of RF transmission and signal delivery, “efficiency improvement” has traditionally been understood to represent increased data rate/throughput, or higher robustness, typically with direct tradeoff between those two parameters .PT-2’s explorations also found that efficiency improvement could be considered in terms of minimization of white spaces within a given transmission band, allowing greater density of usage of the available spectrum in all locations.
- Current spectrum usage policy has uniform spectrum uses across all geographical areas. Can different uses be crafted for different areas (e.g., urban vs. rural) in order to improve efficiency and/or for other improvements?
- Current spectrum usage policy has uniform usage across all broadcast bands. Can different uses be crafted for different bands?
- A critical component of ATSC 3.0 development will be the *transition plan* from ATSC 2.0 required to enable universal broadcaster and consumer adoption.
- Regarding such transition, it is important to consider that available “transitional spectrum” (as in the analog-to-digital transition) will diminish over time as the need for spectrum increases.
- Finally, consider that future broadcast networks may operate as part of a larger “smart” or “adaptive” network. In such a configuration, network bandwidth availability may drive changes in configuration of (for example) essence components and physical layer attributes. Bandwidth may be “throttled” based on usage requirements against business rules yet to be determined. For example, there may be times when television broadcast components are more- or less-compressed (equating to variability of occupied spectrum), or services are added/dropped, based on requirements to increase or decrease available bandwidth for other supported services. Such capability could be managed by a third party, based on some set of negotiated business conditions. To improve interference performance and/or reduce overall spectrum occupancy requirements, integrating and/or synchronously interleaving multiple carriers across adjacent markets also may be helpful.

5.3 Additional Considerations

5.3.1 Timing Considerations

One area not specifically explored by PT-2 was core technology, including future processing power, storage size and speed, and other new device technologies. Transmission and reception will both benefit from the core technology advances predicted by Moore’s Law. Projecting the capabilities and timeframes of future professional and consumer devices will be essential in predicting what may be available and when, and thus what is commercially viable in a next-generation broadcast system.

5.3.2 Business Considerations

The business of broadcasting and how it might change with the introduction of the various technologies was only peripherally considered by PT-2, largely as part of each of the technologies under consideration. The team appropriately focused on the technology as a first step, but consideration of potential changes in the television broadcasting business must be accommodated in the work on ATSC 3.0. One presentation [16] suggested a separation of the delivery platform from the services platform, and suggested that these two platforms could be operated by different entities. Such a business arrangement would be a dramatic departure for current broadcast organizations. If changes of this kind are made, how do broadcast organizations migrate to corresponding new business arrangements? Migration is an essential consideration and must be considered.

Many other distribution platforms represent current competition for viewers and users of the distributed content. A few of the presentations to PT-2 discussed harmonization and merging of services into a larger and more complete service offering. This is another topic that needs further exploration, especially with respect to how it might shape a next-generation broadcast industry.

5.3.3 Regulatory Considerations

PT-2 was not charged with consideration of the regulatory environment. Nevertheless, this is an essential ingredient in the development of any new broadcast system, and has become increasingly important in the current context. Next-generation technologies will of course need to be aligned to future regulations. Therefore this is an area that the ATSC 3.0 process must monitor closely as its work progresses.

5.4 Conclusions of the Planning Team

Historically, the physical layer of television broadcasting has always been operated as a one-way service. Based on presentations at the PT-2 symposia and other inputs, however, a feedback (or return) channel is clearly desirable in any future DTV system, and will likely be a baseline consumer expectation on all television-related devices by the time of ATSC 3.0’s deployment.

An alternative physical layer to provide that two-way connectivity and enable interactive services (expected to be the Internet) is currently gaining interest. Nevertheless, the need to maintain traditional one-way signal transmission over the air will continue to be of predominant importance, and it is critical that such service remain reliable and robust, independent of backchannel connectivity. MIMO/MISO technology appears to be the direction most developers are pursuing to resolve reliability issues while maintaining or improving bandwidth efficiency.

The demand for increased quality and/or quantity of video, audio, and other content/services in television broadcasting is seemingly inexorable. Higher data rates delivered via more spectrally efficient and robust transmission remains a top priority. To accommodate this ongoing change, a flexible next-generation system is essential – one that can continue to grow as technology and demand advance.

One method of achieving such flexibility is the decoupling of the next-generation system’s layers from one another, as has proven effective in accommodating ongoing development for digital networking systems. While this is more difficult to accomplish in a broadcast service, it should be carefully considered, particularly if the next-generation system were to include inherent two-way capability. In any case, evaluation of proposals going forward should favor steps that provide the fewest restrictions to further growth.

Based on the in-depth analysis and extensive inputs from the industry, it is evident to PT-2 that the enabling technologies outlined in this report could provide significant performance improvement in the physical layer, systems and essence coding components of DTV service. A next generation television broadcast system built on an appropriate selection of these technologies could support higher payload, more robust transmission, flexible and interactive services, and increased audio and video quality. These component technologies are either already available or in development.

PT-2 hereby delivers to ATSC these conclusions, along with the results of investigations it has conducted into relevant technologies presented herein, for their use as a foundation to development of appropriate technologies and standards enabling the ATSC 3.0 era.

6 REFERENCES

6.1 ATSC NGBT Symposium Presentations

The following references pertain to presentations made at ATSC's two symposia on Next Generation Broadcast Television. Most of these referenced documents are slide presentations, but in some cases manuscripts are also provided.

All reference resources below are retained on ATSC's "Kavi" document repository, and therefore the associated hyperlinks provided are accessible only by ATSC members.

[1] "Advanced Video Codecs: What's on the Horizon?" Anthony Vetro, Mitsubishi Electric Research Labs, Cambridge, MA. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17667/Anthony-Vetro-Video-Codecs.pdf

[2] "A Revolutionary Digital Broadcasting System: Achieving Maximum Possible Use of Bandwidth." Naohisa Kitazato, Yoshiharu Dewa, Robert Blanchard and Mark Eyer, Sony Corporation, San Diego, CA. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17328/A%20Revolutionary%20Digital%20Broadcasting%20System-%20Sony%2010-7-2010.pdf (manuscript)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17674/Naohisa-Kitazato-Next-Generation-Broadcast.pdf (slides)

[3] "Next-Generation High Efficiency Video Coding (HEVC) Standard." Gary J. Sullivan (Co-chair JCT-VC, Chair VCEG, Co-Chair Video in MPEG), Microsoft, Redmond, WA. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18564/09-2011_02_HPA_NextGenHEVC%20GJS%20d4.pdf

[4] "Transmission Technologies for Next-Generation Digital Terrestrial Broadcasting: Increasing Transmission Capacity toward Super Hi-vision." Kenichi Murayama, Makoto Taguchi, Takuya Shitomi, Hiroyuki Hamazumi and Kazuhiko Shibuya, Japan Broadcasting Corporation (NHK), Tokyo, Japan. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17673/Kenichi-Murayama-NGBT.pdf (slides)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17309/ATSC_NGBT_manuscript_final.docx (manuscript)

[5] "Meeting the Requirements of Next-Generation Broadcast Television Audio." Robert Bleidt (Fraunhofer USA DMT), Harald Fuchs, Stefan Meltzer, Stephan Schreiner (Fraunhofer IIS) and Skip Pizzi (Consultant), Fraunhofer-Gesellschaft, Munich, Germany. ATSC Symposium on Next-Generation Broadcast, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17675/Stefan-Meltzer-Television-Audio.pdf (slides)

http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17322/Fraunhofer_NGBT_Symp_Paper,FINAL.pdf (manuscript)

[6] "Next-Generation 3-D Audio Creation, Transmission and Reproduction." Jean-Marc Jot, DTS, Inc., Calabasas, CA. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18571/07-Jean-Marc_Jot_3D_Audio.pdf

[7] "Latest Trends in Worldwide Digital Terrestrial Broadcasting and Application to the Next Generation Broadcast Television Physical Layer." Lachlan Michael, Makiko Kan, Nabil Muhammad, Hosein Asjadi and Luke Fay, Sony Corporation, San Diego, CA. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17442/Lachlan-Michael-Physical-Layer.pdf (slides)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17310/ATSC_PT2_Sony_PHY_paper.doc (manuscript)

[8] "Beyond Coding: 3D and beyond." James D. Johnston, DTS, Inc., Calabasas, CA. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17670/James-Johnston-Beyond_Coding.pdf

[9] "Information Theory, Shannon Limit and Recent Advances in Error Correction Codes for Terrestrial DTV Broadcasting." Yiyan Wu, Bo Rong and Gilles Gagnon, Communications Research Centre Canada, Ottawa, ON, Canada. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18543/02-PT-2%20Infor%20X-Rec%20FEC%20Yiyan-V10.pdf

[10] "Near-Capacity BICM-ID-SSD, Good for Future DTTB Systems." Jian Song, Tsinghua University DTV Technology R&D Center, Beijing, China. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18545/03-Presentation_JianSong.pdf

[11] "DVB-T2 in relation to the DVB-x2 Family of Standards." Nick Wells (Chairman of DVB TM-T2 working group), BBC R&D, London, England, UK. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18546/04-Wells_ATSC_T2%20v2.pdf (slides)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18560/Wells%20ATSC%20paper%20on%20T2.pdf (manuscript)

[12] "On the Application of MIMO in DVB." Joerg Robert, Institut für Nacrichtentechnik, Technische Universität Braunschweig, Braunschweig, Germany. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18548/05-Joerg_Robert.pdf

[13] "Basic Study of Next-Generation Digital Terrestrial Broadcasting Transmission System for Handheld and Mobile Reception." Yoshikazu Narikiyo, Masahiro Okano and Masayuki Takada, Japan Broadcasting Corporation (NHK) Science & Technology Research Laboratories, Tokyo, Japan. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18570/08-ATSC_sym_presen_Narikiyo(NHK)_final.pdf (slides)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18558/ATSC_sym_paper_Narikiyo(NHK)_final.pdf (manuscript)

[14] "A Hybrid MIMO System for Terrestrial Broadcasting of Next Generation ATSC." Bong Gyun Jo and Dong Seog Han, Wireless Communication Laboratory, Kyungpook National

University, Daegu, Korea. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18550/06-ATSC_20110213_bonggyunJo.pdf

[15] "Toward the Construction of Hybrid-Cast™." Kinji Matsumura, Yasuaki Kanatsugu and Hisakazu Katoh, NHK Science and Technology Research Laboratories, Tokyo, Japan. *ATSC Symposium on Next Generation Broadcast Television,* October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17390/Kinji-Matsumura-Hybridcast.pdf (slides)
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17313/TowardTheConstructionOfHybridcast_NHK.pdf (manuscript)

[16] "Exploring Innovative Opportunities in ATSC Broadcasting: Convergence in the UHF Band in USA." Mike Simon, Rohde & Schwarz, Columbia, MD, and Mark Aitken, Sinclair Broadcast Group, Hunt Valley, MD. *ATSC Symposium on Next Generation Broadcast Television*, February 15, 2011.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/18553/10-Exploring%20Innovative%20Opportunities%20in%20ATSC%20Broadcasting%20MAA%2016x9%20rev3.pdf

[17] "Self-Organizing Broadcast Network." Hyoungsoo Lim and Heung Mook Kim, Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea. *ATSC Symposium on Next Generation Broadcast Television*, October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17312/SON_Paper_101008_Lim.pdf

[18] "Surround Demystified." Christophe Chabanne, Charles Robinson and Jeff Riedmiller, Dolby Laboratories, San Francisco, CA. *ATSC Symposium on Next Generation Broadcast Television,* October 19, 2010.
http://members.atsc.org/apps/org/workgroup/pt-2/download.php/17669/Christophe-Chabanne-Surround-Demystified.pdf

6.2 Other references

6.2.1 PT-3 references

The following references pertain to presentations made to ATSC's Planning Team 3 on Internet Enhanced Television (PT-3). These documents are retained on ATSC's "Kavi" document repository, and therefore the associated hyperlinks provided are directly accessible only by authorized ATSC members.

[19] "HbbTV Status." Presentation by Ludovic Pierre, OpenTV, and Robin Wilson, Nagra to ATSC PT-3, September 2, 2010.
http://members.atsc.org/apps/org/workgroup/pt-3/download.php/16998/HbbTV_ATSC_30Aug10.pdf

[20] "Introduction of MPEG Multimedia Transport." Presentation by Sung Oh Hwang, Samsung Electronics to ATSC PT-3, February 22, 2011.
http://members.atsc.org/apps/org/workgroup/pt-3/download.php/18614/MMT_Introduction_V2.pdf

[21] "Open Hybrid TV Standard (Korea Next Generation Broadcast Forum)". Presentation by Gomer Thomas, LG Electronics to ATSC PT-3, January 24, 2011.
http://members.atsc.org/apps/org/workgroup/pt-3/download.php/18277/KNGBF%20OHTV%20Standard%20r1%200%20(ATSC-PT3).ppt

6.2.2 Further references

The following references are produced by other standards bodies. All should be readily accessible to non-members of those organizations.

[22] "Hybrid Broadcast Broadband TV." ETSI Technical Specification TS 102 796 V1.1.1 (2010-06). European Technical Standards Institute, Sophia Antipolis, France, June 2010.
http://www.etsi.org/deliver/etsi_ts/102700_102799/102796/01.01.01_60/ts_102796v010101p.pdf

[23] "Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Frequency (RF) system scenarios (3GPP TR 36.942 version 9.1.0 Release 9)." ETSI Technical Report TR 136 942 V9.1.0 (2010-10). European Technical Standards Institute, Sophia Antipolis, France, October 2010.
http://www.etsi.org/deliver/etsi_tr/136900_136999/136942/09.01.00_60/tr_136942v090100p.pdf

[24] "EBU Evaluations of Multichannel Audio Codecs." EBU-Tech 3324. European Broadcast Union, Geneva, Switzerland, September 2007.
http://tech.ebu.ch/docs/tech/tech3324.pdf

Annex A: Programs of the ATSC NGBT Symposia

A.1 FIRST SYMPOSIUM (19 OCTOBER 2010)

8:00 a.m.	Continental Breakfast
9:00 a.m.	**Welcome and Introduction** Jim Kutzner, *PBS*
9:15 a.m.	**Advanced Video Codecs: What's On The Horizon?** Anthony Vetro, *Mitsubishi Labs*
10:00 a.m.	**Transmission Technologies for Next-generation Digital Terrestrial Broadcasting** Kenichi Murayama, Makoto Taguchi, Takuya Shitomi, Hiroyuki Hamazumi and Kazuhiko Shibuya, *NHK STRL*
10:45 a.m.	Break
11:15 a.m.	**Latest Trends In Worldwide Digital Terrestrial Broadcasting and Application** Lachlan Michael, Makiko Kan, Nabil Muhammad, Hosein Asjadi and Luke Fay, *Sony Corporation*
11:45 a.m.	**Toward The Construction Of Hybrid-Cast** Kinji Matsumura, Yasuaki Kanatsugu and Hisakazu Katoh, *NHK STRL*
12:15 p.m.	Lunch
1:00 p.m.	**Keynote Speaker** Mark Schubin: "Television Broadcasting's First 125 Years"
1:30 p.m.	**Surround Demystified** Christophe Chabanne, Charles Robinson and Jeff Riedmiller, *Dolby Laboratories*
2:00 p.m.	**A Revolutionary Digital Broadcasting System: Making The Fullest Possible Use Of Bandwidth** Mark Eyer, Naohisa Kitazato, Yoshiharu Dewa and Robert Blanchard, *Sony Corporation*
2:30 p.m.	**Beyond Coding: 3D and Beyond** James D. Johnston, *DTS*
3:00 p.m.	Break
3:30 p.m.	**Self-Organizing Broadcast Network** Hyoungsoo Lim and Heung Mook Kim, *ETRI*
4:00 p.m.	**Meeting the Requirements of Next Generation Broadcast Television Audio**, Stefan Meltzer, Robert Bleidt, Harald Fuchs, and Stephan Schreiner, *Fraunhofer*, and Skip Pizzi, *Consultant*
4:30 p.m.	**Wrap-Up Discussion**
5:00 p.m.	Adjourn

A.2 SECOND SYMPOSIUM (15 FEBRUARY 2011)

8:00 a.m.	Registration Open
9:00 a.m.	Welcoming Remarks: **ATSC Next Generation Broadcast Television** Jerry Whitaker, *ATSC*, and Jim Kutzner, *PBS*
9:15 a.m.	**Information Theory, Shannon Limit and Recent Advances in Error Correction Coding for Terrestrial DTV Broadcasting** Yiyan Wu, Bo Rong and Gilles Gagnon, *CRC*
10:00 a.m.	**Near-Capacity BICM-ID-SSD, a Good Candidate for Future DTTB System** Qiuliang Xie, Kewu Peng, Zhixing Yang and Jian Song, *DTV Technology R&D Center, Tsinghua National Laboratory of Information Science and Technology*
10:30 a.m.	Break
11:00 a.m.	**DVB-T2 in relation to the DVB-x2 Family of Standards** Nick Wells, *BBC Research & Development*
11:30 a.m.	**On the Application of MIMO in DVB** Joerg Robert, *Institut für Nachrichtentechnik, Technische Universität Braunschweig*
12:00 p.m.	Lunch
1:30 p.m.	**A Hybrid MIMO System for Terrestrial Broadcasting of Next Generation ATSC** Jo Bonggyun and D.S. Han, *School of Electronics Engineering, Kyungpook National University, Korea*
2:00 p.m.	**Next-Generation 3-D Audio – Creation, Transmission and Reproduction** Jean-Marc Jot, *DTS*
2:30 p.m.	Break
3:00 p.m.	**Basic study of Next-Generation Digital Terrestrial Broadcasting transmission system for handheld and mobile reception** Yoshikazu Narikiyo, Masahiro Okano and Masayuki Takada, *NHK STRL*
3:30 p.m.	**Next Generation High Efficiency Video Coding Standard** Gary Sullivan, *Microsoft*
4:00 p.m.	**Exploring Innovative Opportunities in ATSC Broadcasting: Convergence in the UHF band in USA** Mike Simon, *Rohde & Schwarz,* and Mark Aitken, *Sinclair Broadcast Group*
4:45 p.m.	**Closing Remarks**
5:00 p.m.	Adjourn

Annex B: Comparison Matrix of Existing DTV Standards

The following table presents a comparison of various attributes of existing DTV formats.

Attribute	Existing Standards										
	ATSC A/53	ATSC A/153	DVB-T	DVB-T2	DVB-H	ISDB-T	DTMB	CMMB	MediaFLO-EV	802.16M	LTE-Advanced
Status	Complete	Complete	Complete	Complete	Complete	Complete	Complete	Complete	Complete	Complete	Complete
Channel Bandwidth, β (MHz)	6	6	6,7,8	1.7,5,6,7,8,10	5, 6, 7, 8	6,7,8	8	2,5,6,7,8	5,6,7,8	5,7,8.75,10,20	1.4,3,5,10,20
Modulation Type	Single Carrier	Single Carrier	Multi-Carrier CP-OFDM	Multi-Carrier CP-OFDM	Multi-Carrier CP-OFDM	Multi-Carrier CP-OFDM	Single//Multi-Carrier PN-OFDM	COFDM	COFDM	OFDMA-DL OFDMA-UL	OFDMA-DL DFTS-OFDM-UL
# of sub-carriers (k=1024)	1	1	2k, 8k	1k, 2k, 4k, 8k, 16k, 32k	2k, 4k, 8k	2k, 4k, 8k	1, 3780	4k	1k, 2k, 4k, 8k	0.5k, 1k, 2k	128, 256, 512, 1k, 2k
Sub-Carrier Modulation	8VSB	8VSB	4,16,64 QAM	4,16,64,256 QAM	4,16,64 QAM	4,16,64 QAM, DQPSK	4,16,32,64 QAM, 4QAM-NR	BPSK, 4,16QAM	QPSK, 16 QAM	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM
Inner FEC	Conv Code	SCCC Code	Conv Code	LDPC	Conv Code	Conv Code	LDPC	LDPC	Turbo	Turbo	Turbo
Outer FEC	Reed Solomon	Reed Solomon	Reed Solomon	BCH	Reed Solomon	Reed Solomon	BCH	RS	None or APP Layer	None or APP Layer	None or APP Layer
Net Bit Rate, R (Mbit/s) in 6 MHz	19.392658	0.0294 to 21.5	3.7 to 31.7	7.4 to 50.3	3.7 to 23.751	3.651 to 23.234	4.81 to 32.49	2.0 – 12.1	2.64 to 10.8	N/A	N/A
AWGN C/N@TOV	15dB	{3.4, 4.5, 7.9}dB	3.5 to 20.2dB	1.0 to 22.0dB	3.6 to 22.2dB	4.9 to 22.0dB	2.24 to 18.3dB	2.7 to 12.0dB	-1.3 to 9.3dB		
TU-6 C/N@TOV					8.5 to 27.0dB						
Pedestrian B C/N@TOV					7.6 to 22dB						
Link Spectral Efficiency, R/β (bit/sec/Hz)	3.2321097	3.23210967	0.62 - 4.0	0.87 - 6.65	0.62 - 4.0	0.61 - 3.87	0.60 - 4.1	0.33 – 2.02	0.44 - 1.8	~0.5-3.0	~0.5 - 3.0
Transport Source Coding	MPEG2	IP	MPEG2 or H.264	MPEG2 or H.264	H.264	MPEG2	MPEG2, H.264 and/or AVS	MPEG2, H.264 and/or AVS	Sync Layer (NAL)	IP	IP
Scalable QoS	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes

	ATSC A/53	ATSC A/153	DVB-T	DVB-T2	DVB-H	ISDB-T	DTMB	CMMB	MediaFLO-EV	802.16M	LTE-Advanced
Coverage Area	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps	F(50,90) Maps		
Service Area	*NOTE: FCC_dtvreport.pdf on page 22										
Peak//Avg Power	6.5dB	6.5dB	8.5 to 12.5dB	8.5 to 12.5dB	8.5 to 12.5dB	8.5 to 12.5dB	6.5 to 12.5dB	8.5 to 12.5dB	8.5 to 12.5dB	8.5 to 12.5dB	8.5 to 12.5dB
PAPR reduction	None	None	None	ACE or TR: <2.0dB	None	None	None	None		None	DFTS-OFDM Uplink Only
Time Interleaving Depth (ms)	4	7.15	0.6 to 3.5	<500 kcells	0.6 to 3.5	0 to 400	200 to 500	n/a[5]	750 min	Small (~1ms)	Small (~1ms)
Phase Noise Tolerance[1]	Good	Good	Average	Average	Average	Average	Average	n/a[5]	Average	Average	Average
Impulse Noise Tolerance[2]	Good	Good	Poor	Good	Poor	Good	Good	n/a[5]	Good	Good	Good
Dynamic Multipath Tolerance[3]	Poor	Good	Good	Good	Good	Good	Good	n/a[5]	Good	Good	Good
Strong Static Multipath Tolerance[4]	Poor	Good	Good	Good	Good	Good	Good	n/a[5]	Good	Good	Good
Unique Feature	Low C/N Requirement	Simulcast HDTV//Mobile DTV		Rotated Constellations, FEF's, Multi-Pipes, MISO		Segmented OFDM	TDS-OFDM, PN Sequence Insertion	High speed mobile reception	Whole Second PHY Structure, Independent Access to Each Stream, Full Stat Mux, Wide and Local	Unicast, Multicast, Broadcast Modes, Mostly Optimized Unicast Bursty Traffic	Unicast, Multicast, Broadcast Modes, Mostly Optimized Unicast Bursty Traffic
Antenna Configuration	1x1	1x1	1x1	1x1, 2x1	1x1	1x1	1x1	1x1	1x1	1x1, 2x2, 4x4,8x8	1x1, 2x2, 4x4, 8x8

NOTES:

1: Phase Noise tolerance at TOV of -80dBc/Hz@20kHz offset = Good; mid -90'sdBc/Hz@20kHz offset is average.

2: Impulse Noise tolerance at TOV of 17-25dB attenuation of D/U is good, 9-14dB attenuation of D/U is poor.

3: Dynamic multipath tolerance of >75Hz Doppler is good, <5Hz Doppler is poor.

4: Strong Static Multipath tolerance of 0dB D/U up to 10usec post is good; < 2dB D/U up to 10usec post is poor.

5: Performance results unavailable at the time of this writing.

Annex C: Spectral efficiency of existing and proposed DTV transmission schemes

The chart below presents the coding performance of all known, existing DTV transmission standards, along with the projected performance of several new proposals investigated by PT-2. (The latter are represented by the last four items on the chart's legend.)